

The Principles of Electronic Communication

EB-551A

TEXTBOOK

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**THE PRINCIPLES OF ELECTRONIC
COMMUNICATION, Textbook**

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INTRODUCTION

In today's world of communications, there is a continuous need for knowledgeable, skilled individuals to design, maintain, and service the communication systems of today and tomorrow.

New technologies are developed every day in the world of communication. These new technologies require a continuous education. This is the third in a series of courses designed to give you a solid foundation on which to base this education.

The Principles of Electronic Communication course will take you through the basics of telephony. These theories are then applied to the telephone equipment as it is used in the communications industry.

In the first section of this course we will introduce you to the important properties of electrical signals, and review frequency, phase, and power fundamentals. We conclude by describing some examples of how these signals are applied in audio and telephone systems. You will learn about the terminology, the quantities, and the relationships of these electrical signal properties.

The second section of this course describes an important aspect of signals that involves power and frequency. This will prepare you to identify the frequencies and power levels of the many signals within a variety of waveforms. You will also learn what distortion is, what causes it, and how to measure it.

The third section of this course describes the results and purposes of processing multiple signals through linear, non-linear, and filter circuits. You will also learn about harmonics and intermodulation products, and how they can be used to your advantage. Your knowledge and understanding of combining signals in both linear and non-linear circuits, and of selecting and suppressing intermod products, is fundamental to the study of communications technology.



In the fourth section of this course you will study the components and circuits that are used to create and amplify audio and telephone signals. The content of this section bridges the gap between the theory fundamentals of Sections One, Two, and Three, and the operation of actual telephone and audio systems and circuits which follow in the next course in this series.

COURSE OBJECTIVES

When you complete this course, you will be able to:

1. List the frequency ranges of audio, voice frequencies, and the divisions of the electromagnetic spectrum.
2. Calculate power gains, losses, and reference power in dB units, and convert these levels to voltage, current, and power ratios.
3. Identify common signal waveforms, and show how they are created by combining multiple frequency sine waves.
4. Specify the common forms of waveform distortion applied to signals as they pass through electronic circuits.
5. Determine the effects of selective filtering on the suppression and/or selection of specific harmonic signals.
6. Identify and calculate the intermodulation frequencies that result from processing two signals through nonlinear circuits.
7. Identify the effects of phase and filtering on the generation and suppression of harmonic and intermod signals.
8. Explain the operating principals of input/output transducers.
9. Explain the characteristics of FETs and ICs.

COURSE OUTLINE

Introduction

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- Objectives

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- Telecommunications Signals

 - Electromagnetic Signals

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 - Wavelength

- Phase

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- Self-Test Review

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Summary
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- Self-Test Review

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- Objectives

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- Combining Amplifier Circuits

 - Signal Levels and Distortion

 - Amplifying Low-Level Signals

 - The Complete Amplifier

- Integrated Circuit Audio Amplifiers

- Passive Components

- Summary

- Self-Test Review

Section 1

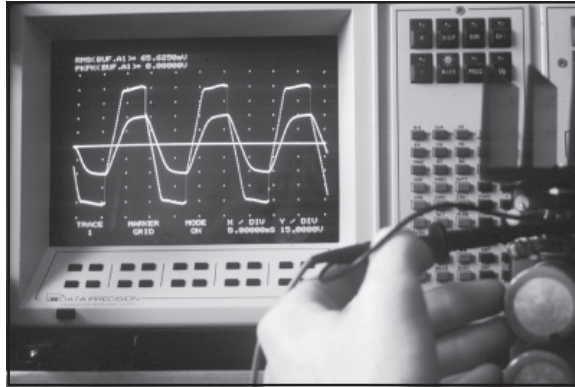
ELECTRICAL SIGNALS

CHAPTER 1 — Signal Characteristics

INTRODUCTION

You were probably first introduced to the term “frequency” when you studied the fundamentals of alternating current (AC) in earlier courses. These introductory presentations of AC generally deal only with sine waves applied to electrical power topics. Electrical signals are also AC, but they are distinct from the AC of electrical power systems. The major distinctions between signals and power include the following:

- A. Signals have many waveforms—they are not always sine waves, like AC power.
- B. Signals are mainly used to transfer information, rather than power.
- C. Signals occupy many frequencies—not just the single frequency of electrical power systems.
- D. Signal information often originates in a form other than electrical energy.



Signals are like AC power, but they vary in frequency and they transfer information.

In their simplest form, telephones are electrical machines. They create electrical signals from acoustic (sound) waves, they transmit and receive the signals, and they convert the electrical signals back to sound waves. To fully understand telephones and telecommunications systems, you must master electrical fundamentals—especially those that relate to signals. Even the most advanced electronic systems will be easier to understand if you have a solid knowledge of signal fundamentals.

OBJECTIVES

When you complete this chapter, you will be able to:

- Define the properties of signals in both acoustic and electrical form.
- Describe and dimension signal frequency, wavelength, and phase.
- Apply the terms and formulas of basic AC to electrical signals.
- List the frequency ranges of audio, voice frequencies, and the divisions of the electromagnetic spectrum.
- Calculate the relationships of time and frequency, phase and distance, and phase and time.

ACOUSTIC AND ELECTRIC AUDIO SIGNALS

In acoustics, frequency is used to describe the pitch of a sound—the number of vibrations-per-second of your eardrum, and of the object producing the sound. For example, a high-pitched sound from a shrill whistle vibrates your eardrum at a high frequency—many vibrations per-second; while a low-pitched sound from a tuba vibrates your eardrum at a low frequency. Figure 1-1 shows the frequency ranges of speech and popular musical instruments within the limits of human hearing. You should observe that the human voice primarily occupies frequencies within the range of about 70 Hz to 3000 Hz. Pipe organs and electronic music synthesizers have the greatest range of sound output frequencies within the range of human hearing. The frequency range, specified to be within the limits of human hearing, is from 20 Hz to 20 kHz. Signals within this range are often called “audio frequency signals.”

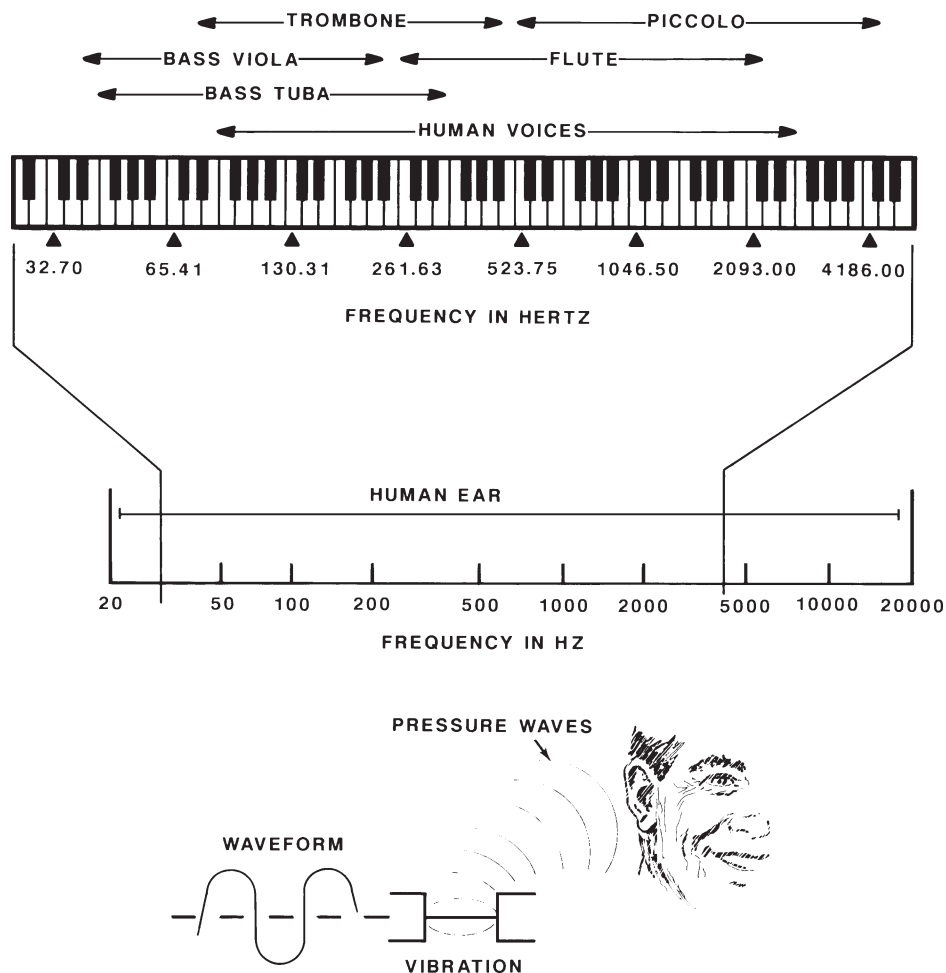


Figure 1-1

Frequency range of sound signals.

Telephone and electronic sound systems are designed to process acoustic signals. To process these signals they must first be converted from their acoustic form to an electrical form. After electronic processing, the signals may be converted back to an acoustical form. The general term for the device that converts from acoustical-to-electrical, and/or electrical-to-acoustical, signals is the transducer.

Telephone and audio systems use transducers to convert acoustic signals to and from electric signals.

Telephone handset transmitter and receiver elements (mouthpieces and earpieces), microphones, and loudspeakers are examples of popular transducers. The best microphones and loudspeakers, such as those used for the recording and reproduction of music, have the ability to convert the widest range of frequencies. This quality is specified as the transducer's frequency response.

When signals are in an electrical form, their frequency response extends far beyond the range of acoustic sound signals. These electrical signals are also distinct from acoustic signals because they don't vibrate the air around them. Electrical signals are represented by voltage and current variations within conductive circuits. What "vibrates" are the electrons moving back and forth along a wire, or voltage variations acting upon an electronic component.

TELECOMMUNICATIONS SIGNALS

Alternating currents serve many purposes in addition to AC power, they play a key role in electronic communications. The familiar services that we use on a daily basis depend upon alternating current *electrical signals* within specific frequency ranges. Among these services are audio, video, telephone, coded data, radio, television, radar, navigation, and control. When these electrical services are specifically used to communicate information, they are classified under the general technical field called "telecommunications." Telecommunications signals are the alternating current signals used by the telecommunications services to transfer information.

Telecommunications signals are AC signals used to communicate information.

In earlier times, the technical fields were divided into specialties. Instead of studying “electronics,” for example, you would learn a specific field, such as “telephone,” or “radio,” or “sound systems.” There were also divisions within these communications fields that separated them into “wired” and “wireless” specialties. Modern technology has brought aspects of all of these fields together, unifying them under the broader category of “telecommunications” technology.

Electromagnetic Signals

When electrical currents flow back and forth along conductive circuit paths, they produce changing magnetic and electric fields—electromagnetic fields—within the air, or space, that surrounds those paths. For example, radio antennas generate electromagnetic fields to transmit radio signals. These changing electromagnetic fields, now in the form of radio waves, will then regenerate a current flow when they intercept another conductive circuit path—a radio receiver’s antenna. The word “radio” actually evolved from the more scientific term “electromagnetic radiation.”

Radio Waves are electromagnetic energy that allows us to communicate without wires.

Frequency Ranges

By international agreement, telecommunications signals are strictly classified and regulated. Through regulation, it is possible to assure worldwide communication with a minimum of unwanted interference. In general the full range of frequencies are classified in decades ($\times 10$) as the frequency bands of the electromagnetic spectrum. These frequency bands are divided as shown in Table 1-1.

You may recognize some of the band designations of the spectrum by their titles used for broadcast television—VHF and UHF. It is within these bands that the frequencies of the VHF and UHF TV channels fall.

Table 1-1

Frequency Range	Metric Subdivision	Adjectival Designation	
30 to 300 hertz	Megametric waves	ELF	Extremely low frequency
300 to 3000 hertz		VF	Voice Frequency
3 to 30 kilohertz	Myriametric waves	VLF	Very low frequency
30 to 300 kilohertz	Kilometric waves	LF	Low frequency
300 to 3000 kilohertz	Hectometric waves	MF	Medium frequency
3 to 30 megahertz	Decametric waves	HF	High frequency
30 to 300 megahertz	Metric waves	VHF	Very high frequency
300 to 3000 megahertz	Decimetric waves	UHF	Ultra high frequency
3 to 30 gigahertz	Centimetric waves	SHF	Super high frequency
30 to 300 gigahertz	Millimetric waves	EHF	Extremely high frequency
300 to 3000 gigahertz or 3 terahertz	Decimillimetric waves		

For example, conventional AM broadcast radio services fall within the frequencies of the MF band, from 0.54 to 1.7 MHz (540 to 1700 kHz); while FM (and FM stereo) broadcast radio services fall within the VHF band from 88 to 108 MHz. International shortwave radio operates within the frequencies of the HF band; while most radar and microwave relay and satellite systems operate within the frequencies of the SHF band. Telephone voice signals operate over the full range of the VF band from 300 Hz to 3 kHz. (The VF telephone signals are combined in telephone systems to also occupy frequencies in other bands.)

DIMENSIONS

Electrical signals and radiated electromagnetic waves have specific characteristics within each band of frequencies. For example, the lower frequency bands have very long waves that follow the earth's curvature when radiated from antennas; while the higher frequency bands have proportionally shorter waves that allow sharp focusing of radiated signals. Within circuits and equipment, wiring paths for the lower frequency signals are not critical; but circuit paths, and the size of components, become extremely critical at the higher frequencies where wavelengths are short.

Signals are cyclic and repetitive. They are generated in the form of waves that alternate, like waves in the sea. When we describe “wavelength,” we refer to the distance that a wave occupies (or moves) within one of its “cycles.” One word used to describe part of a wave’s cycle is “phase.” Phase is the angular relationship between current and voltage in alternating-current circuits.

A simple way to see how the dimensions of an alternating signal relate to each other is to compare these dimensions to the characteristics of a bicycle wheel. If you mark the edge of a bicycle wheel where it touches the ground, and then move the bicycle a distance of one complete rotation of the wheel, as shown in Figure 1-2, you can visualize the terms that are applied to signals; as follows:

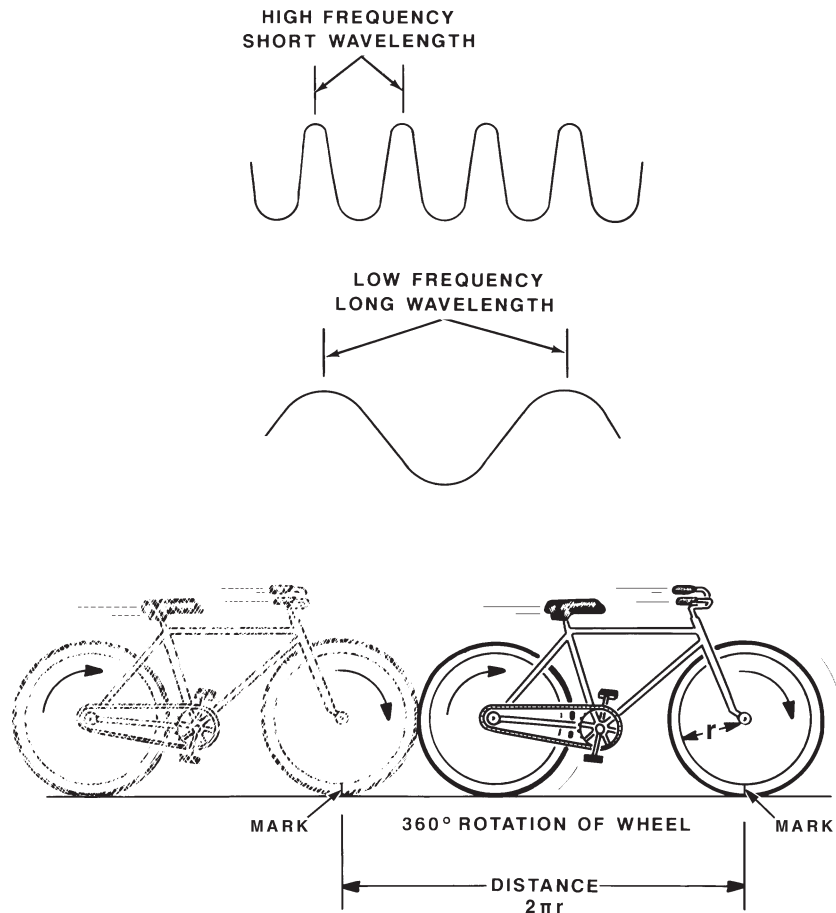


Figure 1-2

Signal frequency vs. wavelength.

- A. One complete **cycle** is 360 degrees.
- B. One **wavelength** is the distance moved in completing one cycle.
- C. One **radian** is like the radius of the wheel, 2π radians is like the circumference of the wheel, which is the distance of one wavelength (360 degrees).
- D. **Frequency** is the number of cycles that occur in a time period.
- E. **Velocity** is the number of wavelengths that occur in a time period.
- F. **Phase** is that portion of rotation that occurs within one cycle.

As you proceed through the study of signal fundamentals, it may be helpful to remember this example of a bicycle wheel, and its movement along a road, to understand the dimensions of signals.

Wavelength

A wavelength is the distance traveled by a wave in the time required for one complete cycle.

The wavelength of specific frequency signals can be determined by simple calculation. In general, the wavelength of any signal is:

$$\text{wavelength} = \frac{\text{velocity}}{\text{frequency}}$$

For acoustic signals in air, their velocity is about 1100 feet-per second (330 meters-per-sec). Therefore, the wavelength of an acoustic sound of 550 Hz is calculated as:

$$\text{wavelength} = \frac{\text{velocity}}{\text{frequency}}$$

$$\text{wavelength} = \frac{1100 \text{ ft/s}}{550 \text{ Hz}} = 2 \text{ ft}$$

or

$$\text{wavelength} = \frac{330 \text{ m/s}}{550 \text{ Hz}} = 0.6 \text{ m} = 60 \text{ cm}$$

Electrical signals travel at about the speed of light. Their velocity is specified as 984×10^6 feet-per-second (300×10^6 meters-per-sec). The wavelength of a 550 Hz electrical audio signal is calculated as:

$$\text{wavelength} = \frac{\text{velocity}}{\text{frequency}}$$

$$\text{wavelength} = \frac{984 \times 10^6 \text{ ft/s}}{550 \text{ Hz}} = 1.79 \times 10^6 \text{ ft}$$

or

$$\text{wavelength} = \frac{300 \times 10^6 \text{ m/s}}{550 \text{ Hz}} = 0.55 \times 10^6 \text{ m}$$

Using this same formula, increasing the 550 Hz frequency by 10, to 5500 Hz, results in a wavelength 1/10th that of the 550 Hz signal; a wavelength of 0.179×10^6 feet (55×10^3 meters). By contrast, a UHF signal operating at 984 MHz will have a wavelength of just 1 foot (0.3 meters).

PHASE

One complete cycle of a signal is specified in units of phase as 360° . The physical length of one full wave of this signal—one wavelength—is also one cycle long. Wavelength is sometimes measured in units of radians instead of degrees. One radian (abbreviated rad) is 57.3° , π (3.14) radians are half a wavelength (180°). One wavelength is 2π (6.28) radians (360°). Figure 1-3 shows one cycle of a signal dimensioned in units typically applied to measurements of wavelength and phase—fractions, degrees, and radians.

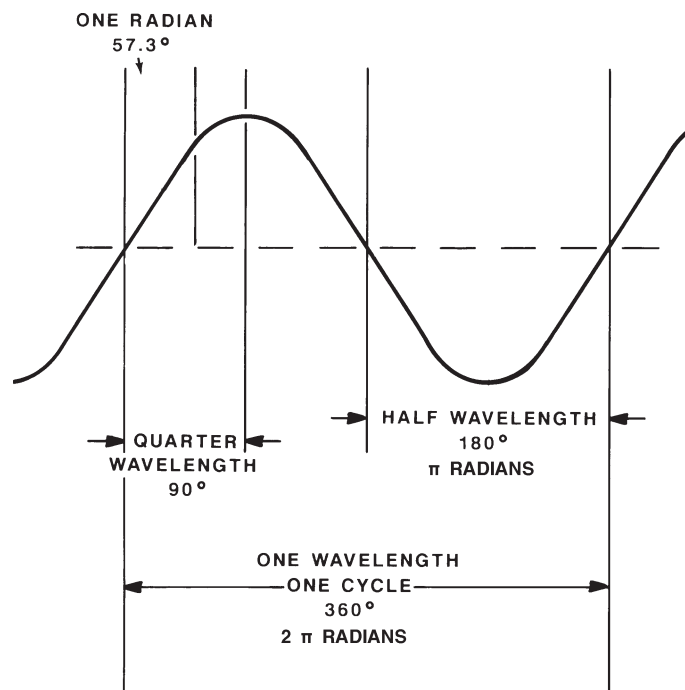


Figure 1-3

Signal wavelength and phase.

With shorter wavelength signals there is a difference in phase along their signal paths. With very short wavelength signals there will be a phase shift even within circuit components, such as resistors. These shifts in phase often prevent components from functioning as they should. Similar problems occur with longer wavelength signals when they must travel over longer circuit paths like cable communication links, longer-length audio cables, and telephone lines. Phase problems are usually anticipated when circuit paths extend beyond .05 wavelengths (18° or 0.3 rads).

An important thing to keep in mind when you work with wavelengths is the units in which these quantities are specified and measured. Frequency is measured in *units of time*; usually in seconds. Wavelength is measured in *units of distance* usually in feet or meters, etc. Phase is measured in *units of degrees* ("phase angles"), or in *fractions or decimal parts of a wavelength*.

Frequency is measured in units of time, wavelength is measured in units of distance, and phase is measured in degrees or in parts of a wavelength.

Frequency, time, wavelength, phase, and distance are all interrelated; it's easy to become confused when you are first learning the relationships between these factors. One source of confusion is the oscilloscope display, which indicates time-per-cycle along its X-axis. When you visualize wavelength by thinking of one cycle of a signal on an oscilloscope display, don't forget that it's not a true representation of wavelength you are looking at. Wavelengths are measured in units of distance!

Frequency vs. Time

Frequency, measured in Hz, identifies the number of electrical cycles that occur in one second of time. Time, in relation to frequency, is measured in seconds, and identifies the period (elapsed time) per cycle of a signal's frequency.

You can see that there is a basic reciprocal relationship between frequency and time, as you probably know from your use of the timebase calibration of the oscilloscope. The formulas are:

$$\text{frequency} = \frac{1}{\text{time}} \text{ and time} = \frac{1}{\text{frequency}}$$

$$f = \frac{1}{t} \qquad t = \frac{1}{f}$$

Knowing frequency, you can calculate time; and knowing time, you can calculate frequency. For example, you can compute the frequency of a signal with a time period of 2 msec. per cycle, as:

$$f = \frac{1}{t} = \frac{1}{2 \text{ ms}} = \frac{1}{(2 \times 10^{-3} \text{ s})} = 500 \text{ Hz}$$

And you can compute the time period of a 2 kHz signal as:

$$\begin{aligned} t &= \frac{1}{f} = \frac{1}{2 \text{ kHz}} = \frac{1}{(2 \times 10^3 \text{ Hz})} \\ &= 5 \times 10^{-4} \text{ s} = 0.5 \text{ ms} \end{aligned}$$

Time And Frequency Of Electrical Networks

Many electrical circuits, including filters and tuners, use inductance and capacitance (LC) to obtain resonance—the response of a circuit to certain frequencies. This is illustrated in Figure 1-4. The time required for signal current to cycle in these circuits is important in determining the frequency at which they operate. This time can be computed using the following formula:

$$t' = \sqrt{LC}$$

Here, t' is the time period for one radian of phase shift in seconds, L is the network inductance in henrys, and C is the network capacitance in farads. This circuit application is the basis for measuring the wavelength, or cycle of a signal, in rads.

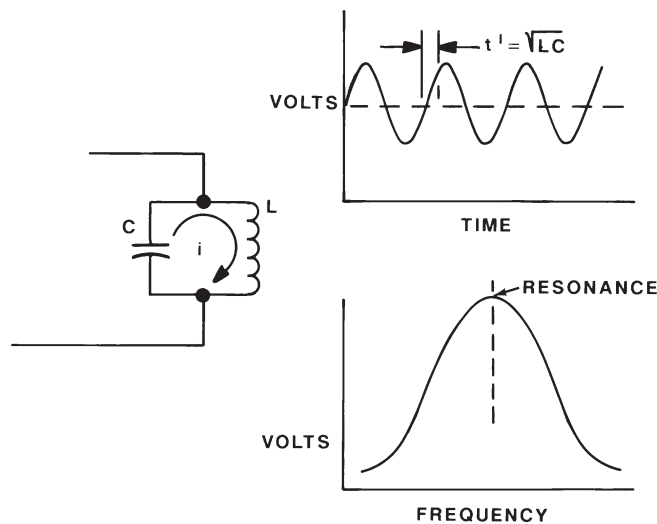


Figure 1-4

Resonant LC network.

You can calculate the time period for one radian of a resonant LC network with 2 mH of inductance, and 5 μ F of capacitance, as:

$$t' = \sqrt{LC} = \sqrt{2 \text{ mH} \times 5 \mu\text{F}}$$

$$t' = \sqrt{(2 \times 10^{-3} \text{ H})(5 \times 10^{-6} \text{ F})}$$

$$t' = 1 \times 10^{-4} \text{ s} = 100 \mu\text{s}$$

The result of this calculation computes the time of one rad (radian) of phase shift for the signal within the electrical network. Thus the time-per-cycle is computed by multiplying t' by 2π ($2\pi \text{ rads} = 360^\circ$):

$$t = 2\pi t'$$

$$t = 2\pi \sqrt{LC}$$

The frequency of the resonant network is the reciprocal of that value:

$$f = \frac{1}{2\pi \sqrt{LC}}$$

For the earlier LC network example, with 2 mH of inductance and 5 μ F of capacitance, the time period of one rad was 100 μ s. We can use this value to calculate the time period of one cycle, and to calculate the frequency of the signal, as follows:

$$t = 2\pi t' = 2\pi \times 100 \mu\text{s} = 628 \mu\text{s}$$

$$f = \frac{1}{t} = \frac{1}{(2 \times 3.14)(100 \times 10^{-6} \text{ s})}$$

$$f = \frac{1}{628 \times 10^{-6} \text{ s}} = 1592 \text{ Hz}$$

The LC network is resonant at a frequency of 1592 Hz.

NOISE IN SIGNALS

Throughout this chapter we have described the characteristics of acoustic and electric signals that are primarily used for communication. Just as there are signals that allow for communications systems to function, there are other, undesirable signals that prevent reliable communications with these systems. Noise is probably the most prominent of these undesirable signals. Noise prevents reliable communication by masking telecommunications signals.

The effects of acoustic noise and electrical noise are similar. Think of attempting a conversation with someone in a noisy machine shop, or while riding in a noisy vehicle. The interference of random background noise forces you to shout to be heard, and soft-spoken words go unheard. Another example of noise masking is how distant sounds can be heard in the quiet forest, but not on the streets of a busy city. Like these examples of interfering “clatter,” electrical noise within telecommunications signals reduces reliability of communication, requires higher signal power levels, and reduces a receiver’s ability to pick up weak or distant signals.

Electrical noise can occur at a single frequency, or at a number of simultaneous frequencies. However, in a strict sense noise is defined as random in frequency, and distributed over a range of frequencies. Noise is the “hiss” you hear in the background of radio and telephone speech, and it’s the “snow” you see in the background of television pictures.

Electrical noise is defined as random in frequency, and distributed over a range of frequencies.

Every electrical circuit, and even the airspace through which telecommunications signals pass, adds noise to the signals. As the signals pass through more circuits, they get weaker; while the noise level increases. At the point where the signal levels equal the noise level, communications systems don’t work. It is for this reason that noise is one of the most critical factors in telecommunications systems; and that noise reduction is a high priority among people who work in telecommunications technology.

SUMMARY

In this chapter we discussed the major distinctions between signals and power. Signals have many frequencies, while AC power is a single frequency sine wave. Signals are used to transfer information. Signal information often originates as acoustic (sound) energy. A transducer is used to convert acoustical signals to electrical signals, and vice versa.

When electrical signals communicate information, they are classified as “telecommunications signals.” Telecommunications signals are normally transmitted as electromagnetic radio waves. Radio wave frequencies are strictly regulated to assure communication with a minimum of interference.

Signals are cyclic and repetitive and can be mathematically expressed in cycles, wavelengths, radians, frequencies, velocities, and phases.

Noise is an undesirable signal that can be either acoustic or electrical. Electrical noise can occur at one or more frequencies, and can be added to signals by electrical circuits, or when they travel through the air.

SELF-TEST REVIEW

1. What is the most important difference between AC as applied to power, and AC as applied to signals?
 - A. The voltage level.
 - B. The current level.
 - C. The power level.
 - D. The frequency.
2. What is the most important difference between acoustic and electric information signals?
 - A. Their frequency.
 - B. Their power level.
 - C. Their audibility.
 - D. Their noise level.
3. The electrical component that converts signals from acoustic-to-electric, or electric-to-acoustic form is called a _____ .

4. The technical field that uses electrical signals to communicate information is called _____ .
5. The shortened form of the term “electromagnetic radiation” is _____ .
6. Which band of the electromagnetic spectrum contains the frequencies for FM stereo broadcast?
 - A. MF.
 - B. HF.
 - C. VHF.
 - D. UHF.
7. Which band of the electromagnetic spectrum contains the frequencies for AM broadcast radio?
 - A. MF.
 - B. HF.
 - C. VHF.
 - D. UHF.
8. Which band of the electromagnetic spectrum has signals with the shortest wavelengths?
 - A. EHF.
 - B. ELF.
 - C. VHF.
 - D. VLF.
9. What is the approximate velocity for acoustic waves in air at sea level? _____ ft/s or _____ m/s
10. What is the approximate velocity for electrical waves in wires? _____ ft/s or _____ m/s
11. The dimensions for the phase of half of a signal wavelength are: _____ cycle _____ degrees _____ rads
12. The unit measure of frequency is _____ .
13. The unit measure of wavelength is _____ .

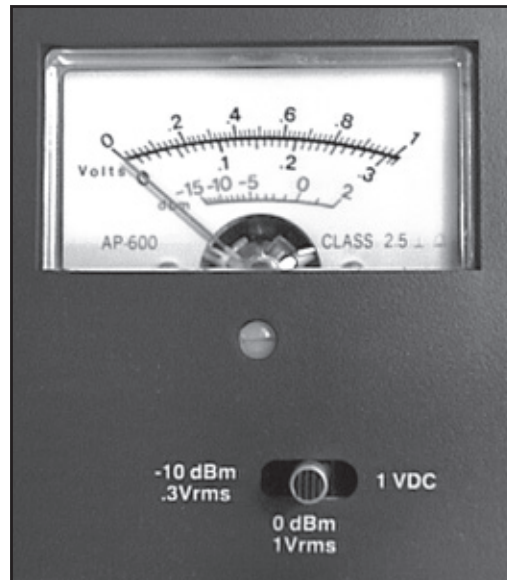
14. Phase is measured in _____ .
15. When will electrical noise interfere most with communications signals?
 - A. When the noise is near the signal's frequency.
 - B. When the noise is near the signal's power level.
 - C. When the noise is distributed over a range of frequencies.
 - D. When the noise is at varying power levels.
16. What is the wavelength of a 3 kHz electrical signal? _____ ft
or _____ m
17. What is the frequency of a signal with a time period of 0.5 ms-per-cycle? _____ Hz
18. What is the time period-per-cycle of a 20 kHz signal? _____ μs
19. What is the time period for one radian of a resonant electrical network with 0.5 mH inductance, and 2 μF capacitance? _____ μs
20. What is the time period-per-cycle of a resonant electrical network with 1 mH inductance, and 2 μF capacitance? _____ μs
21. What is the frequency of a resonant electric network with 5 mH inductance, and 1 μF capacitance? _____ Hz

CHAPTER 2 — Signal Power

INTRODUCTION

There are many aspects of power to know about for telecommunications technology. There are simple power ratings, like those printed on light bulbs, which are based upon the applied voltage and current consumed by an electrical component. From your earlier studies, you may remember that true power may be less than apparent power, depending upon the relative phase of a component's voltage and current.

You should be aware that true power, rms power, requires that AC signals have sine waveforms, with voltage and current levels specified in rms units. You should also know that the actual power levels of non sine-wave signals can't be simply calculated by multiplying voltage times current. These, and other factors, complicate this important property of electrical signals—power.



True Signal Power is rms volts $\times$ rms amps,
with signal voltage and current in-phase.

In telecommunications, power is used to specify many aspects of signals and equipment. Signals are rated in units of power level, while equipment (also a component) is rated in units of power consumption. Equipment, components, and circuits also affect a signal's power level with their properties of power gain or power loss. To complicate things even further, almost all signal power levels, and almost all circuit gain and loss values, are specified in logarithmic dB units. So, while it doesn't take much knowledge of electrical power to select a light bulb, there is much to know about power applied to telephone and audio systems in telecommunications practice.

OBJECTIVES

When you complete this chapter, you will be able to:

- Apply the terms and formulas of basic AC to electrical signals.
- Use basic terminology to describe and dimension signal power.

SINE-WAVE SIGNAL POWER

In this section, we'll review the fundamentals of sine-wave signal power, using the relationships you likely learned in your earlier studies of AC. Keep in mind the distinctions emphasized earlier in Chapter 1: signals are like AC power, but they vary in frequency, and they are primarily used to transfer information—not to provide power to equipment. Therefore, when we specify power levels for signals, we often specify them in two dimensions—power vs. frequency.

The following formulas are often used for signal power specification and measurement. They assume that the signals are *sine waves*, and that the voltage and current ratings are in rms units.

The three common formulas for calculating power are:

$$P = V \times I \quad P = \frac{V^2}{R} \quad P = I^2 R$$

For signals, the formulas that are most often used are:

$$P = V \times I \quad P = \frac{(V_{rms})^2}{R} \quad P = \frac{(V_{p-p})^2}{8R}$$

NOTE: Peak-to-Peak voltage values for sine waves are 2.828 times their rms equivalents. 2.828 squared is 8; so the calculations of power from p-p values are reduced by this factor of 8.

The formulas, in their transposed form, are:

$$V_{rms} = \sqrt{P \times R} \text{ and } V_{p-p} = \sqrt{P \times 8R}$$

The formulas using voltage and resistance data are used more often than those that include current values, because signal current is often impractical to measure. By contrast, test points for signal voltage and resistance are usually accessible and convenient to measure. Also, both rms and p-p voltages are used in power calculations, due to the form in which measured data are available. Signals are either measured with oscilloscopes, or with meters. The scope displays p-p signal voltages, while meters are usually calibrated for rms voltages.

You'll likely also use the formulas with voltage and resistance data when doing signal power calculations, due to the controllable variables of Ohm's Law. It's the voltage and resistance (impedance) values that are most-often directly adjustable in electrical circuits; current and power levels are a consequence of voltage and resistance.

Signal voltage and circuit resistance are the controlling quantities, while current and power are the resulting quantities.

Practical Examples

Following are four practical examples of calculating voltage and power. Figure 2-1 shows the circuit for each example.

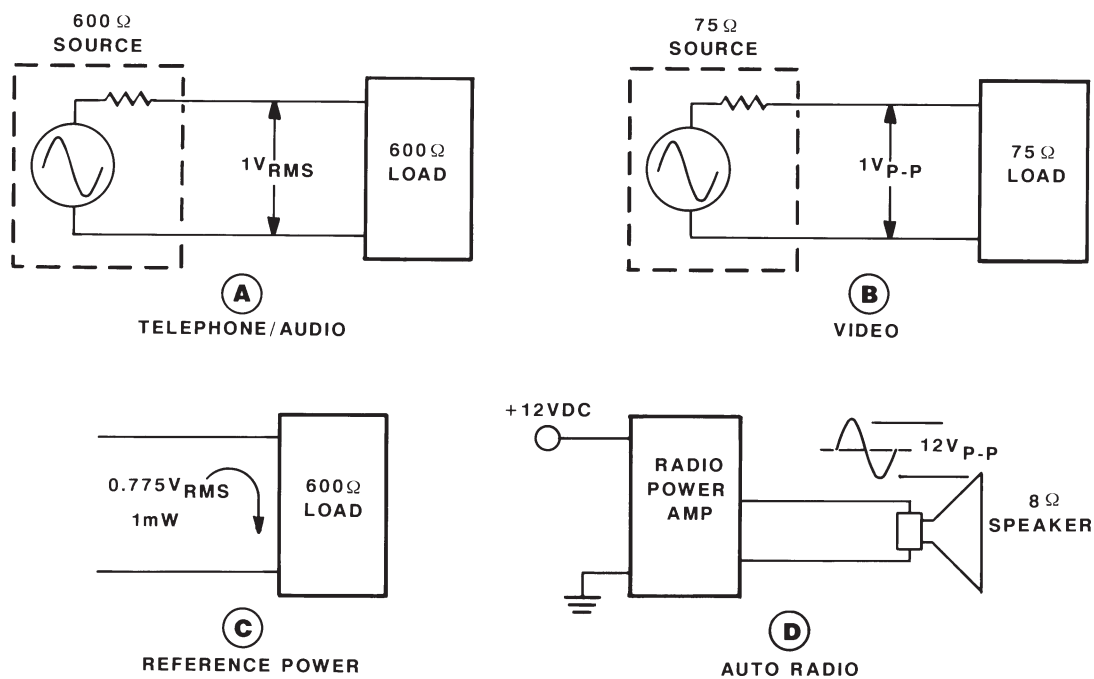


Figure 2-1
Circuit examples.

A typical resistance (impedance) value for telephone and audio systems, as shown in Figure 2-1A, is 600 ohms. For reference, the power level of a 1-volt rms signal across 600 ohms is:

$$P = \frac{(V_{\text{rms}})^2}{R} = \frac{(1 \text{ V})^2}{600} = 1.67 \times 10^{-3} = 1.67 \text{ mW}$$

The typical output level specified for video signal equipment, as shown in Figure 2-1B, is 1-volt p-p at 75 ohms. The power level of these signals is:

$$P = \frac{(V_{\text{p-p}})^2}{8R} = \frac{(1 \text{ V})^2}{(8 \times 75 \Omega)} = 1.67 \text{ mW}$$

Most audio systems specify a reference power of 1.0 milliwatt at 600 ohms, as shown in Figure 2-1C. The rms and p-p voltage levels of the signal required for this reference power are:

$$V_{\text{rms}} = \sqrt{P \times R}$$

$$V_{\text{rms}} = \sqrt{1 \times 10^{-3} \text{ W} \times 8 \times 600 \Omega}$$

$$V_{\text{rms}} = 0.775 V_{\text{rms}}$$

$$V_{\text{p-p}} = \sqrt{P \times 8R}$$

$$V_{\text{p-p}} = \sqrt{1 \times 10^{-3} \text{ W} \times 8 \times 600 \Omega}$$

$$V_{\text{p-p}} = 2.19$$

With the limitation of a 12-volt DC power supply, the signal swing available to drive an 8-ohm loudspeaker in an automobile radio is 12 volts p-p, as shown in Figure 2-1D. Thus the maximum power at the speaker will be:

$$P = \frac{(V_{\text{p-p}})^2}{8R} = \frac{(12 \text{ V})^2}{(8 \times 8 \Omega)} = 2.25$$

When power boosters are used for car radio and stereo systems, they increase the radio's DC power supply voltage in order to obtain the higher speaker power levels they advertise.

These examples are intended to give you the knowledge necessary to determine signal power levels, and to develop a sense of judgment about signal voltage and power levels. With this knowledge, and some skill developed through practice, you should be able to make quick calculations to evaluate power specifications.

POWER IN NETWORKS AND CIRCUITS

The true power of an electrical network is defined as that power dissipated (converted to heat energy) as a result of current flowing through the network. The formula, $P = V \times I$, is accurate only if the applied voltage and current of the network are in-phase. To the extent that network voltage and current are out of phase, the true power dissipated becomes less than the multiple of $V \times I$.

Power Factor

The factor by which the power is reduced is called the “power factor.” Power factor is computed as the cosine of the phase angle difference between the current and voltage of an electrical network. The network’s impedance determines this phase angle. If the impedance is completely resistive (with zero inductance or capacitance), the voltage and current will be in-phase at 0° . If the impedance is reactive resistive and inductive or capacitive, the voltage and current will be out-of-phase. The phase angle between the voltage and current is equivalent to the phase angle of the impedance. (Reactive impedances are also often called “complex impedances”.)

The power factor can be calculated, based upon the values of the voltage and current, but reduced by the cosine of the phase angle of the network components’ impedance. In a simple network the calculation procedure is:

$$\text{Power Factor} = \cos \angle \theta = \left[\tan^{-1} \left(\frac{X}{R} \right) \right]$$

In the above equation the power factor equals the cosine of the phase angle ($\angle \theta$ “theta”) of the network impedance. The phase angle equals the arctangent (inverse tangent) of the network reactance (X) divided by the network resistance (R). This calculation is for a single fre-

quency. A completely resistive network will have 0° phase shift. The cosine of 0° is 1.0, so the true power will be 100% of that available to the network. By contrast, a completely reactive (zero resistance) network will have 90° phase shift. The cosine of 90° is 0.0, so the true power will be 0% of that available to the network. These concepts are shown in Figure 2-2.

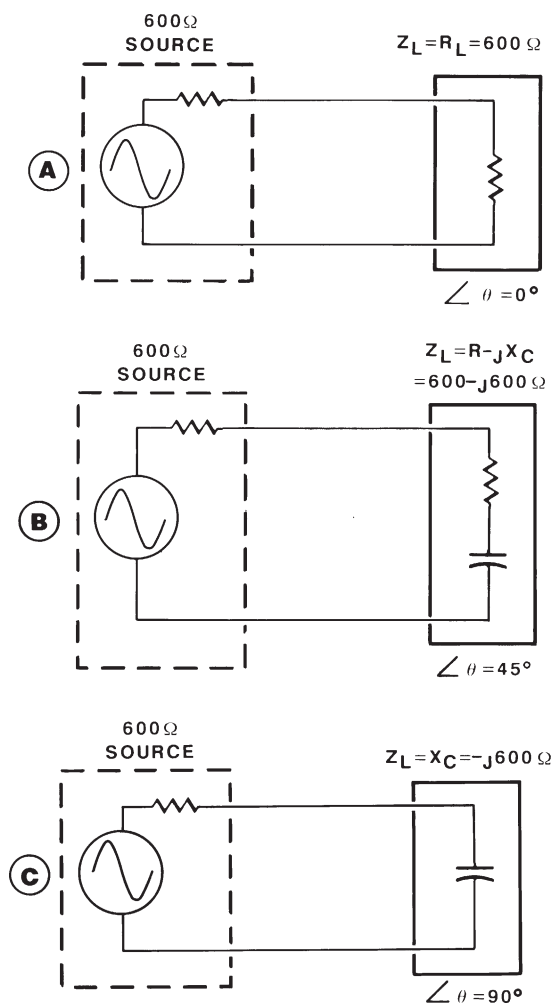


Figure 2-2

Resistive and reactive networks.

For example, the power factor applied to a signal passing along a simple network with 600 ohms resistance, and 600 ohms reactance can be calculated as:

$$\text{Power Factor} = \cos [\tan^{-1} (\frac{600 \Omega}{600 \Omega})] = 0.71$$

For this example, the true power will be reduced to 0.71, or 71% of the multiple of the network's signal voltage and current that would be applied to a purely resistive network (600 ohms resistance for this example).

NETWORK CALCULATIONS

The true signal power for reactive (complex) electrical networks takes the power factor into account when the phase angles of the networks are calculated. The true power is that proportion of applied power that is dissipated in the resistance part of the RL or RC network. The true signal power may be calculated using the following formulas:

$$P = \frac{V^2}{Z \angle \theta} = \frac{V^2}{\left(\frac{X}{\sin \left[\tan^{-1} \left(\frac{X}{R} \right) \right]} \right)}$$

The mathematics may seem a little complicated, but the process of calculation is straightforward. Given that network resistance and reactance are known (and/or the reactance has been calculated for the signal frequency), the three-step process is as follows:

1. Divide the network reactance (X) by the network resistance (R), and obtain the arctangent (inverse tangent) for that result. The resulting angle ($\angle \theta$ "theta") is the phase angle of network impedance, and also the phase angle of the network's signal voltage and current. *Record this angle.*
2. Divide the network reactance (X) by the sine of the phase angle (calculated in part 1 above), to obtain the true network impedance. *Record this value.*
3. Divide the square of the applied signal voltage by the true network impedance (calculated in part 2 above), to obtain the true network power. *Record this as your final value.*

This process will give you accurate answers and a good sense of judgment about network signal power.

Now, let's apply this three-step process to the solution of true signal power for a series reactive network of 600 ohms resistance, 400 ohms reactance, and an rms signal of 8 volts rms, as shown in Figure 2-3.

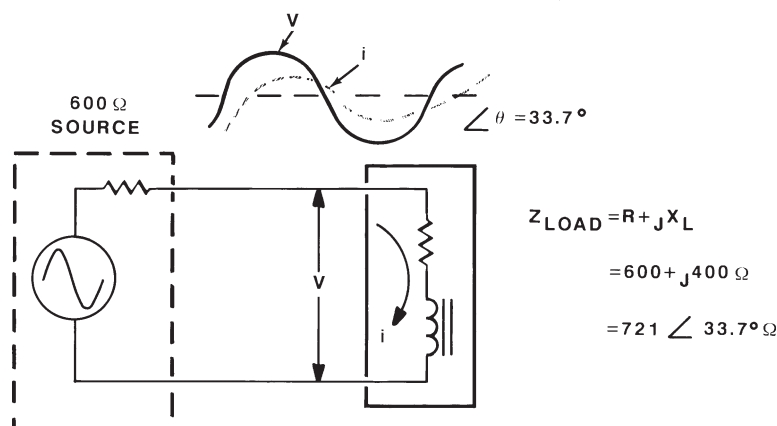


Figure 2-3

Power of a reactive network.

$$\text{Step 1. } \angle \theta = \tan^{-1}\left(\frac{400 \, \Omega}{600 \, \Omega}\right) = 33.7^\circ$$

$$\text{Step 2. } Z = \frac{400 \, \Omega}{\sin 33.7^\circ} = 721 \, \Omega$$

$$\text{Step 3. } P = \frac{(8V_{rms})^2}{721} = 88.8 \, \text{mW}$$

We can also use the simpler power factor approach to calculate the true signal power for the same network. With this approach, we first assume that the circuit is purely resistive at 600 ohms (neglecting the reactance); and then proceed as follows:

$$P = \frac{V^2}{R \cos \angle \theta}$$

$$P = \frac{(8V)^2}{600 \, \Omega} \cos 33.7^\circ$$

$$P = 88.8 \, \text{mW}$$

The power factor approach is simpler, as it can be accomplished in two steps:

1. Determine phase angle.
2. Determine true power.

The impedance approach requires three calculation steps, but it also provides impedance values which cannot be obtained using the power factor approach.

POWER GAINS AND LOSSES

Telecommunication circuits and systems impart gains and losses upon the signals they process. Power losses usually occur as signals pass from one point to another through either wired or wireless circuit paths. Amplifiers are often placed within such circuits to increase signal power levels, and to compensate for the path losses. Amplifiers are also used in conjunction with transducers to increase their signal power levels. The general term used to describe what happens to signals as they move along and through circuit paths is “transmission.” The accumulated power losses or gains within such circuits are called “transmission loss” and “transmission gain”.

As you learn about transmission, you should keep in mind that power gains and losses are not power levels. Gains and losses are *proportional*, and they are specified as the ratio of two signal power levels. When dealing with signals, it is important to maintain a clear distinction between these *proportional gains and losses* and the *absolute power levels* of signals.

One way to keep the distinctions clear is to remember that gains and losses are the ratios of two signal power levels. As such, gains and losses are measured as factors—not Watts or volts. For example, an amplifier may have a voltage gain of 5 (5:1), or a power gain of 100 (100:1). A transmission circuit (circuit path) may have a power loss of 8 (1:8).

Power gains and losses are specified by factors. Power levels are specified in units—watts.

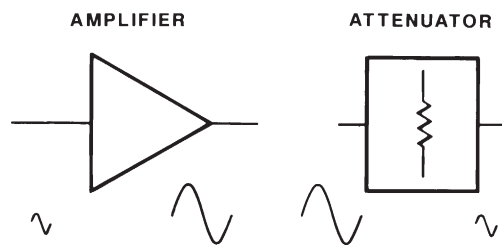
Measurements of transmission gains and losses are made by comparing the power (or voltage) level of one signal to another. For example, the power at the output of an amplifier, compared to the power at its input, is the amplifier's gain; it is the measurement of power ratio. Power ratios are also sometimes called "relative power." In contrast, actual power levels are classified as "absolute power." Table 2-1 may help you keep these classifications separate.

Table 2-1

	Absolute	Relative
Designation	Power Level	Gain or loss
Units		
Arithmetic	WATTS	Ratios or factors
Logarithmic	dbm	dB
Application	Signals	Components and circuits

Power (voltage and/or current) gains in equipment circuits are called "amplification," while power losses are called "attenuation." Amplifiers are designed into systems to provide gain to signals, and to increase the signal's power level. If a 2 milliwatt signal is applied to the input of an amplifier, which then raises the signal power to 10 milliwatts at its output, the amplifier has a power gain of 5 (5:1). (Keep in mind the distinctions between power level and power gain.)

Attenuation is often an undesirable, but accepted, factor that identifies the extent of loss that results from signals passing through systems and circuits. There are also many instances where attenuation is purposely applied to signals as they pass through circuits and equipment. The components used to apply this loss are called "attenuators." Attenuators are used extensively in telephone, audio, and other telecommunications equipment. They are used to reduce signal levels by exact amounts, and they are used for the indirect measurement and control of signal power. Figure 2-4 illustrates the concept of amplification and attenuation.

**Figure 2-4**

Amplifier and attenuator.

SUMMARY

Telecommunications signals are rated in units of power level, while equipment is rated in units of power consumption. Signal power levels are specified in logarithmic dB units. The true power of an electrical network is defined as that power dissipated as a result of current flowing through the network. The factor by which power is reduced is called “power factor.” The accumulated power loss or gain within a telecommunications circuit is called “transmission loss” and “transmission gain.”

SELF-TEST REVIEW

1. When multiplying rms volts $\times$ rms amps, with the signal voltage and current in phase, the resulting product is said to be:
 - A. Apparent Power.
 - B. True Power.
 - C. Rated Power.
 - D. Calculated Power.
2. Identify how power ratings are used to define the operation of components and signals, by writing “1” for components, and “2” for signals next to the following:
 - ___ A. power consumption.
 - ___ B. power level.
 - ___ C. power gain.
 - ___ D. power loss.

3. Which of the following is the most practical data for calculating signal power?
 - A. Voltage and resistance.
 - B. Voltage and current.
 - C. Current and resistance.
 - D. Resistance and inductance.
4. Which impedance is most often used for audio signal transmission circuits?
 - A. 50 ohms
 - B. 75 ohms
 - C. 300 ohms
 - D. 600 ohms
5. How do car radio power boosters increase signal power output to the speakers?
 - A. They reduce speaker impedance below 8 ohms.
 - B. They increase amplifier circuit gain.
 - C. They increase power supply voltage above 12V.
 - D. They use output transistors with higher power ratings.
6. By what approximate percentage will true power be reduced when a signal is applied to an RC network with equal values of reactance and resistance?
 - A. 10%
 - B. 30%
 - C. 50%
 - D. 80%
7. Where is a large portion of true power dissipated in a circuit?
 - A. A resistive network.
 - B. A resistive-capacitive (RC) network.
 - C. A resistive-inductive (RL) network.
 - D. A reactive network.
8. How are power gains and losses specified?
 - A. In watts.
 - B. In volts.
 - C. In ohms.
 - D. In factors.

9. What is the power dissipated in an 8 ohm loudspeaker, with an 8 volt p-p signal across its terminals? _____ W
10. What is the signal power when 8 volts rms is measured across a 600 ohm transmission circuit? _____ mW
11. What p-p signal voltage is required for a reference power level of 1 milliwatt on a 600 ohm audio circuit? _____ V
12. What rms signal voltage is required to drive an 8 ohm loudspeaker at a power level of 50 watts? _____ V
13. What is the power factor of a network with 300 ohms resistance and 200 ohms reactance? _____ What is this value in terms of percentage? _____ %
14. With 8 volts rms applied to an RL network of 100 ohms resistance and 100 ohms reactance, what is the true power dissipated by the network? _____ mW

CHAPTER 3 — Logarithmic Power Ratios

INTRODUCTION

Almost everything in our physical world is logarithmic in character. Figure 3-1 illustrates the idea that when you look down railroad tracks, you see the first ties relatively far apart, but as you look further down the tracks, the ties seem closer together. In fact all of the rail ties are about equally spaced. This visualization of perspective is one example of logarithmic ratios. Signal power gains and losses are also logarithmic ratios, and we perceive the changes in power ratios like visual perspectives.

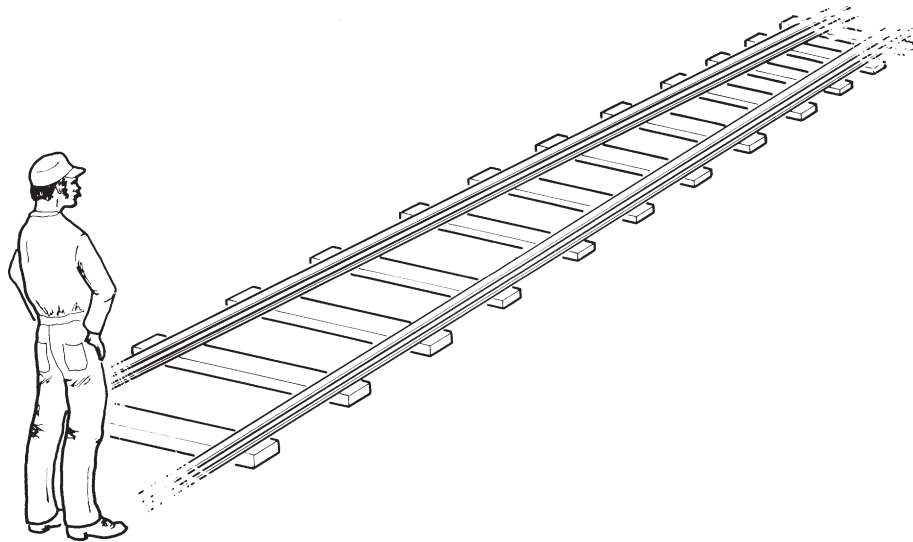


Figure 3-1

Visual perspective of railroad tracks.

We also hear sounds in logarithmic ratios. With audio equipment, for example, the actual difference in sound output from a 50-watt amplifier, compared to a 100-watt amplifier, is barely perceptible. Why pay twice as much for the 100-watt amplifier if you can't hear the difference? In fact it takes a difference of ten times the power, not twice the power, for sounds to register as significantly louder to our ears. To produce sounds significantly louder than a 50-watt amplifier, you will need a 500-watt amplifier!

Like the example with the audio equipment, all acoustical and electrical signals must be scaled in logarithmic units to accurately evaluate their power ratios. The logarithmic ratios of signals are most-often specified in decibel (dB) units.

Signal power ratios are most accurately proportioned when they are scaled in decibel units.

The power ratios of electrical signals are scaled in logarithmic decimal (dB) units.

Table 3-1 shows the relationship between power ratios and voltage ratios, compared to dB units.

Table 3-1

POWER RATIO	DECIBELS (dB)	VOLTAGE OR CURRENT RATIO
1	0	1
1.26	1	1.12
1.58	2	1.26
2.00	3	1.41
2.51	4	1.59
3.16	5	1.78
3.98	6	1.99
5.01	7	2.24
6.31	8	2.51
7.94	9	2.82
10	10	3.16
100	20	10
1 000	30	31.6
10 000	40	100
100 000	50	316
1 000 000	60	1 000
10 000 000	70	3 160
100 000 000	80	10 000
1 000 000 000	90	31 600
10 000 000 000	100	100 000

OBJECTIVES

When you complete this chapter, you will be able to:

- Calculate power gains and losses in dB units, and convert these levels to voltage, current, and power ratios.
- Calculate referenced power levels in dBm, and convert these levels to voltage, current, and power ratings.

CALCULATING DECIBELS

Arithmetic power ratios can be converted to logarithmic decibel units using the following formula:

$$\text{dB} = 10 \log_{10} \left(\frac{P_2}{P_1} \right)$$

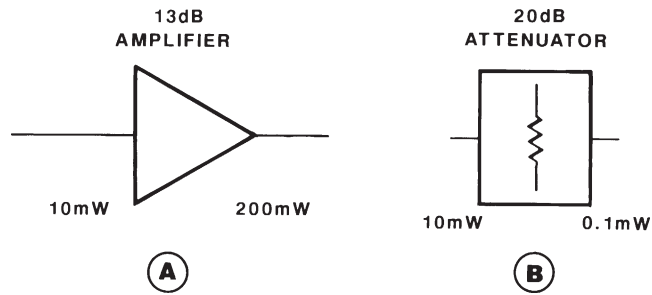


Figure 3-2
Gain and Loss in dB

P_2 and P_1 are the power ratios of two signals, such as those at the output and input of an amplifier. For example, let's consider an amplifier that has a power gain of 20, with 10 mW at its input and 200 mW at its output, as shown in Figure 3-2A. Its signal power gain will be accurately scaled by calculation of its dB value as:

$$\text{dB} = 10 \log_{10} \left(\frac{P_2}{P_1} \right)$$

$$\text{dB} = 10 \log_{10} \left(\frac{200 \text{ mW}}{10 \text{ mW}} \right)$$

$$\text{dB} = 10 \log_{10} 20 = 13 \text{ dB}$$

Keep in mind that the gain of the amplifier will remain the same even if the input signal power level is changed. The output signal power will always be 20 times the input signal power, or 13 dB greater than the input signal power.

Even though actual power levels change, dB values of gain and loss do not change.

An attenuator that causes a signal loss of 100 (its output is 1/100th its input power), as shown in Figure 3-2B, is accurately specified in dB units as:

$$\text{dB} = 10 \log_{10} \left(\frac{P_2}{P_1} \right)$$

$$\text{dB} = 20 \log_{10} \left(\frac{1}{100} \right)$$

$$\text{dB} = 10 \log_{10} 0.01 = -20 \text{ dB}$$

NOTE: The minus (–) sign is often omitted from the specification of attenuation and signal loss, because the words “attenuation” and “loss” imply lower output power. For the above calculation 20 dB of attenuation, and –20 dB mean the same thing.

QUICK dB CONVERSIONS

For convenience, dB values are often determined without resorting to calculation. People who use dB values regularly can quickly convert power ratios to dB (and dB to power ratios) by applying two simple relationships:

1. A power ratio of 2:1 is equivalent to just about 3 dB; thus every time power is doubled, the ratio increases by 3 dB. A 4:1 (2×2) power ratio is 6 dB (3 dB + 3 dB); an 8:1 ($2 \times 2 \times 2$) power ratio is 9 dB (3 dB + 3 dB + 3 dB); and so on. When a reduction in power, or attenuation, is determined, the ratios are simply interchanged: a 3 dB loss yields a power ratio of 1:2, or half the power; a 6 dB loss yields a power ratio of 1:4, or one-fourth the power; and so on.
2. A power ratio of 10:1 is equivalent to exactly 10 dB; thus every time power is increased by a multiple of 10, the ratio increases by 10 dB. Thus, a 100:1 power ratio is 20 dB (10 dB + 10 dB); a 1000:1 power ratio is 30 dB (10 dB + 10 dB + 10 dB); and so on. Again, when a reduction in power, or attenuation, is determined, the ratios are interchanged. In this case, a 10 dB loss yields a power ratio of 1:10, or one-tenth the power; a 20 dB loss yields a power ratio of 1:100, or one-hundredth the power; and so on.

With practice you can become proficient at using these relationships, without using a calculator, to quickly estimate power ratios and dB ratios.

Voltage (or Current) Ratios

Decibels are used to honestly scale signal power ratios. However, signal power is not always convenient to measure directly. Most meters, oscilloscopes, and other instruments indicate voltage, rather than power. In a given circuit, a voltage or current increase will cause a proportionately equal increase in power; so dBs are often used to conveniently scale measured signal voltage (or current) ratios. Keep in mind, however, that these dB conversions are correct *only when the impedances at the points of voltage measurement are the same!*

In practice, signal voltage measurements are often taken at the same terminal of a circuit, as signal levels are adjusted elsewhere in the circuit. In other cases, signal voltages are measured in transmission circuits which maintain matched impedances. In such cases, voltage measurements are a valid means of obtaining direct power gain and loss data in dB.

Signal power scaling in dB units can be obtained from voltage ratios, provided that the impedances are constant.

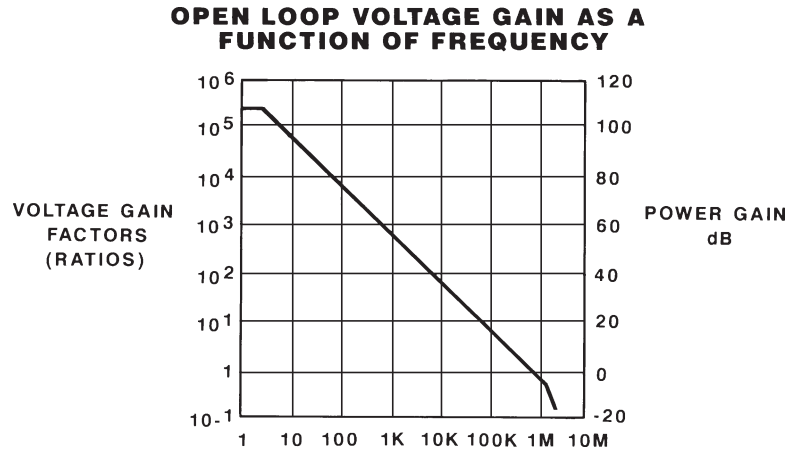
Given the conditions of identical impedances at the points of signal ratios, you can calculate dB values using the following formulas:

$$\text{dB} = 20 \log_{10} \left(\frac{V_2}{V_1} \right)$$

or

$$\text{dB} = 20 \log_{10} \left(\frac{I_2}{I_1} \right)$$

Calculations using these formulas are most often applied to measured voltage ratios, and/or measurements of different signal voltage levels at a single point in a circuit. Figure 3-3 shows a graph of voltage gain vs. frequency response for a typical IC amplifier, with both voltage ratios and dBs scaled to describe its gain. This graph is accurate only if the output impedance (load) on the IC is the same for all of the frequencies where the voltage measurements are taken.

**Figure 3-3**

Frequency vs. dB (power gain) and voltage gain.

At the point on the graph where the voltage gain is 10 (10:1), the power gain is computed as:

$$\text{dB} = 20 \log_{10} \left(\frac{V_2}{V_1} \right)$$

$$\text{dB} = 20 \log_{10} \left(\frac{10}{1} \right)$$

$$\text{dB} = 20 \log_{10} 10 = 20 \text{ dB}$$

In this instance, V_1 is the reference from which signal voltage gain is measured. Remember that dBs are a measure of power gain or signal power ratios, even if the power ratios are obtained indirectly from signal voltage or current ratios!

dBs are a measure of signal power gain, even when they are calculated from voltage ratios.

REFERENCED ABSOLUTE POWER

dB values are used extensively in telephone, audio, and telecommunication practice because they accurately scale signal power ratios. The dB values have an additional advantage for tracing signals and their power levels through circuits and systems; adding and subtracting dB values is the equivalent of multiplying and dividing signal power ratios. This feature of dB scaling makes quick work of evaluating signals and equipment.

Scaling Power In dBm

What remains is to extend the convenience of dB scaling to absolute signal power levels. This conversion is accomplished by referencing absolute signal power levels to some standard level, against which any other power level can be compared for conversion to a dB ratio. In telecommunications systems, the most common standard for absolute power reference is the milliwatt. A 1-milliwatt signal is defined as exactly equal to the reference level, so its power ratio to the reference is zero dB.

A 1.0 milliwatt signal is scaled at 0 dBm; 0 dBm = zero dB in reference to one milliwatt.

Referenced absolute signal power levels can be converted from power levels scaled in milliwatts using the following extension of the dB formula:

$$\text{dBm} = 10 \log_{10} (P_{\text{mW}})$$

A 20 mW signal can be scaled in dBm, as:

$$\text{dBm} = 10 \log_{10} (P_{\text{mW}})$$

$$\text{dBm} = 10 \log_{10} 20 \text{ mw}$$

$$\text{dBm} = 13 \text{ dBm}$$

Practical Examples

We can demonstrate the convenience of the dB/dBm techniques by applying them to a number of practical problems, whose networks are shown in Figure 3-4. First, the gain of an amplifier (or loss of an attenuator) can be quickly determined by comparing the power levels of the signals at the component's input and output.

If the input signal power to a circuit is 3 dBm, and its output signal power is 26 dBm, as shown in Figure 3-4A, the circuit's gain is the difference (or ratio) of the two power levels:

$$26 \text{ dBm} - 3 \text{ dBm} = 23 \text{ dB}$$

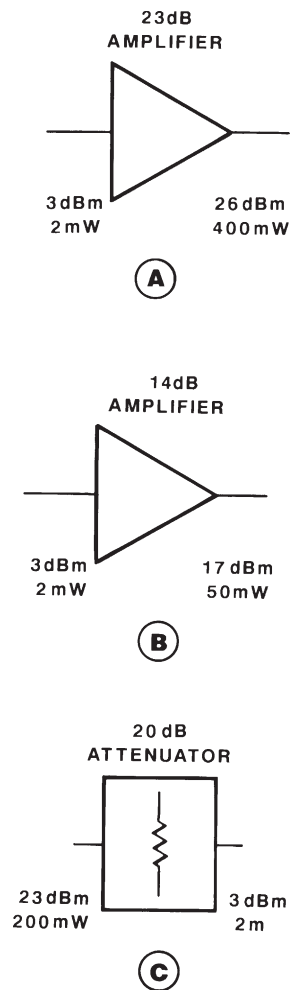


Figure 3-4

Circuit gain and power levels.

The circuit gain is 23 dB. Remember gain and loss are specified in dB, not dBm; and signal power is specified in dBm not dB.

If the input signal power to a circuit is 3 dBm, and its gain is known to be 14 dB, as shown in Figure 3-4B, the circuit's output signal power is simply the sum of the signal level and the gain scaled in dB units:

$$3 \text{ dBm} + 14 \text{ dB} = 17 \text{ dBm}$$

The circuit's output signal power is 17 dBm. Again, keep signal levels, and gain and loss, distinct.

If the output signal level of a circuit is 3 dBm, and the circuit's loss is known to be 20 dB, as shown in Figure 3-4C, the circuit's input signal power is the sum of the output signal level and the circuit loss, as scaled in dB units:

$$3 \text{ dBm} + 20 \text{ dB} = 23 \text{ dBm}$$

The circuit's input signal level of 23 dBm is reduced to 3 dBm by the circuit path loss of 20 dB.

Reverse Calculations

Power and voltage ratios, and absolute power levels, can be obtained by inverting the dB and dBm formulas. The following practical examples will summarize the more common of these calculations.

- A. If you want to obtain power ratios (P_2/P_1) from dB values, use the inverted form of the formula:

$$\text{dB} = 10 \log_{10} \left(\frac{P_2}{P_1} \right)$$

$$\text{thus: } \left(\frac{P_2}{P_1} \right) = \log_{-1} \left(\frac{\text{dB}}{10} \right)$$

The power ratio that yields 6 dB of gain, for example, can be calculated as:

$$\left(\frac{P_2}{P_1} \right) = \log_{-1} \left(\frac{6 \text{ dB}}{10} \right) = 3.98 \text{ (or 3.98:1)}$$

- B. If you want to obtain voltage ratios (V_2/V_1) from dB values, use the inverted form of the formula, but remember that the impedances must be constant:

$$\text{dB} = 20 \log_{10} \left(\frac{V_2}{V_1} \right)$$

$$\text{thus: } \left(\frac{V_2}{V_1} \right) = \log_{-1} \left(\frac{\text{dB}}{20} \right)$$

The voltage ratio that yields 6 dB of gain, for example, can be calculated as:

$$\left(\frac{V_2}{V_1} \right) = \log_{-1} \left(\frac{6 \text{ dB}}{20} \right) = 2 \text{ (or 2:1)}$$

- C. If you want to obtain absolute power levels in milliwatts from dBm values, use the inverted formula:

$$\text{dBm} = 10 \log_{10} P_{\text{mW}}$$

$$\text{thus: } P_{\text{mW}} = \log_{-1} \left(\frac{\text{dBm}}{10} \right)$$

The power level of a 6 dBm signal can be computed as:

$$P_{\text{mW}} = \log_{-1} \left(\frac{6 \text{ dBm}}{10} \right) = 3.98 \text{ mW}$$

Similarly, dB can be referenced to watts (dBw) or femtowatts (dBf). These can be calculated using the same formula as you would use for dBm, the difference being whether you use watts or femtowatts as a reference.

POWER SCALING WITH/PREFIXES

Telephone, audio, and most telecommunications systems operate over a wide range of signal frequencies and power levels. In regard to power, the signal levels may be high in the kilowatt (10^3 watt) range for radio transmitters, and they may be very low in the femtowatt (10^{-15} watt) range for radio receivers. The signals in telephone and audio systems also have a wide dynamic range. For example, certain telephone noise power levels are in the picowatt (10^{-12} watt) range, while audio loudspeaker amplifiers operate in the watts (tens and hundreds of watts) range.

People who work regularly with signals have developed the skill to translate power prefixes to and from referenced (dBm) power levels. They can also quickly trace signal paths through circuits and systems; and they can scale signal power levels by the gains and losses encountered as the signals pass through the systems. The techniques used to conveniently translate power prefixes, and account for gains and losses, are easy to learn. With practice, your skill will develop to perform these translations quickly and accurately.

One key to translating power levels is the Scaled Power Ruler Line shown in Figure 3-5. The ruler line is divided into major prefixes above the line, each spaced at three decimal points apart. These prefixes are referenced to the watt as a “unity” (10^0) value. Below the prefixes are the exponents, in scientific notation, for each of the prefix values from Megawatts to femtowatts. Below the ruler line are the referenced power levels in dBm that correspond to the prefixes above the line. Note that the dBm reference is the milliwatt (10^{-3} watts). Keeping this reference (watt vs. milliwatt) distinction in mind will eliminate difficulties in using the ruler line. The minor divisions between each of the prefixes along the line are divided into factors of 10, each equal to one decimal place, and each equal to 10 dB.

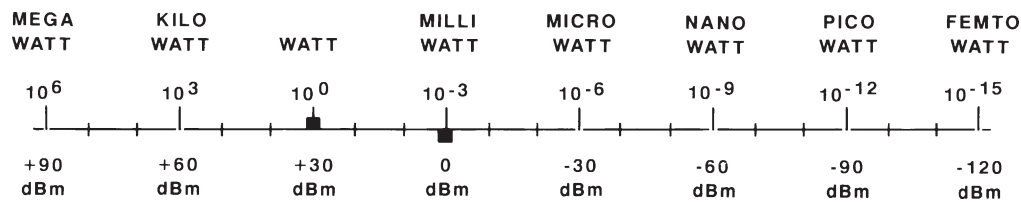
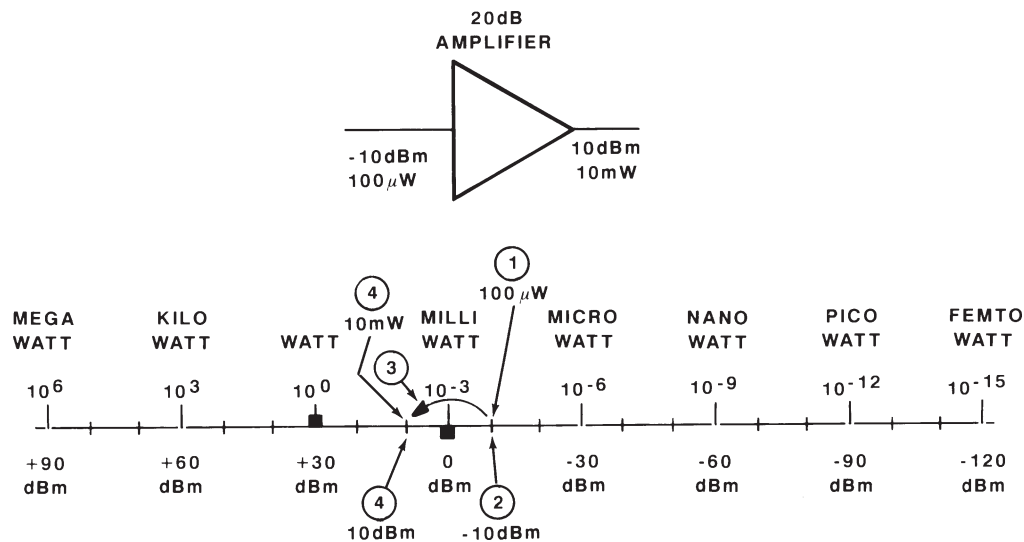


Figure 3-5
The scaled power ruler line.

**Figure 3-6**

Output power of 20 dB amplifier.

Let's apply the ruler line to a simple circuit example, where the input signal power to an amplifier is 100 microwatts ($100 \mu\text{W}$), and the amplifier gain is 20 dB, as shown in Figure 3-6. You can quickly determine the amplifier's output power level, in dBm and milliwatts, using the ruler line (per Figure 3-5) as follows:

1. Locate the signal power level of $100 \mu\text{W}$ at two minor divisions (two decimal points) to the left of the μW prefix mark.
2. Read the signal power level in dBm units. The signal power level is 10 dB smaller (one decimal mark to the right) than 0 dBm. The referenced power level of a $100 \mu\text{W}$ signal is -10 dBm.
3. Add the gain of the amplifier to the input power level by moving two minor divisions (10 dB + 10 dB) to the left.
4. Read the signal power in dBm units. The signal power level is 10 dB larger (one decimal mark to the left) than 0 dBm. The referenced power level of the output signal is 10 dBm.
5. Read the signal power in milliwatts equivalent to 10 dBm, as one minor division (one decimal point) to the left of the mW prefix mark. The output power level is 10 milliwatts.

With the ruler line, you can quickly determine the loss in dB that results when a 1-watt signal is reduced to 100 nanowatts (nW). There are seven major and minor divisions between 1 watt and 100 nW. The loss is 70 dB. The 1 watt signal is at 30 dBm, and the 100 nW signal is at -40 dBm.

Initially, the ruler line can be used most effectively in evaluating signals that are scaled in 10 dB increments. However, with practice in applying the 3 dB = 2:1, and 10 dB = 10:1, factors, you will be able to closely estimate almost all dB, dBm, and watts values without relying on a calculator.

As an example, let's consider an amplifier input signal of -43 dBm. In Figure 3-7, you can see that -40 dBm is 100 nW, -3 dB more is half (1:2) that power level; so -43 dBm is 50 nW.

Now, let's amplify this signal level by 36 dB: 36 dB above -43 dBm (+36, -43) is -7 dBm.

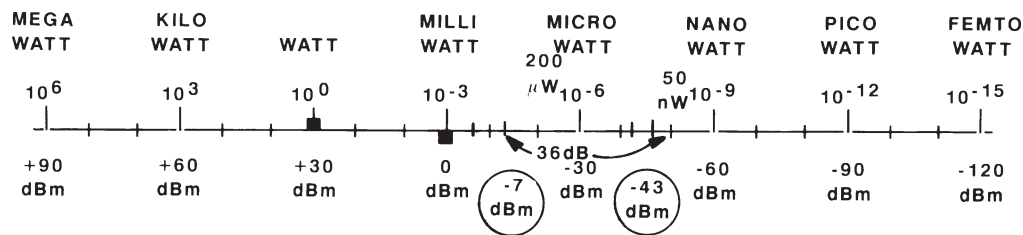


Figure 3-7

Output power of 36 dB amplifier.

Finally let's determine the power level at the amplifier's output: -7 dBm is 3 dB (2:1) above -10 dBm. -10 dBm is 100 μ W, and 3 dB above (2 $\times$) 100 μ W is 200 μ W.

SUMMARY

Signal power ratios are scaled in logarithmic decibel units. A power ratio of 2:1 is equivalent to 3 dB; therefore, every time power is doubled, the dB units increase by 3 dB. A power ratio of 10:1 is equivalent to 10 dB; therefore, when power is increased by a multiple of 10, the dB units increase by 10 dB.

SELF-TEST REVIEW

1. Which of the following is not logarithmic when its level of intensity is specified?
 - A. Sound signals.
 - B. Light energy.
 - C. Electrical signals.
 - D. Monetary exchange.

2. Which difference in dB sound level is considered to be just perceptible?
 - A. 3 dB
 - B. 10 dB
 - C. 30 dB
 - D. 60 dB

3. Which difference in dB sound level is considered to be significantly louder?
 - A. 3 dB
 - B. 10 dB
 - C. 30 dB
 - D. 60 dB

4. Which difference in dB sound level represents a power ratio of one million to one?
 - A. 3 db
 - B. 10 dB
 - C. 30 dB
 - D. 60 dB

5. An amplifier with 13 dB gain has a 2 mW signal applied to its input. What is the signal power level at the amplifier's output?
 - A. 13 mW
 - B. 20 mW
 - C. 26 mW
 - D. 40 mW

6. An attenuator has a signal power level of 2 mW at its input and 0.1 mW (100 μ W) at its output. What is its attenuation factor?
 - A. 3 dB
 - B. 10 dB
 - C. 13 dB
 - D. 20 dB
7. What is the equivalent power ratio for a gain of 19 dB?
 - A. 8:1
 - B. 18:1
 - C. 19:1
 - D. 80:1
8. What is the equivalent gain for a power ratio of 40:1?
 - A. 6 dB
 - B. 16 dB
 - C. 26 dB
 - D. 40 dB
9. What is the equivalent power gain for an amplifier with matched impedances and a voltage ratio of 100:1?
 - A. 10 dB
 - B. 20 dB
 - C. 40 dB
 - D. 100 dB
10. A 2 mW signal passes through a 20 dB amplifier, and then through a 3 dB attenuator. What is the signal power at the attenuator's output?
 - A. 20 mW
 - B. 50 mW
 - C. 100 mW
 - D. 200 mW
11. What is the referenced power level of a 2 mW signal?
 - A. 1 dBm
 - B. 2 dBm
 - C. 3 dBm
 - D. 4 dBm

12. What is the power level of a 13 dBm signal?
 - A. 10 mW
 - B. 20 mW
 - C. 40 mW
 - D. 100 mW
13. A 6 dBm signal is amplified by 10 dB. What is the signal level at the amplifier's output?
 - A. 10 mW
 - B. 20 mW
 - C. 40 mW
 - D. 100 mW
14. A 100 pW signal is equivalent to a power level of:
 - A. -70 dBm
 - B. -80 dBm
 - C. -90 dBm
 - D. -100 dBm
15. A signal at -50 dBm is equivalent to a power level of:
 - A. 10 fW
 - B. 10 pW
 - C. 10 nW
 - D. 10 μ W
16. What is the calculated power gain of an amplifier that increases a 2 mW signal at its input to 500 mW at its output? _____ dB
17. What is the calculated signal loss of an attenuator that reduces a 2 mW signal at its input to 0.5 μ W (500 nW) at its output? _____ dB
18. With matched impedances, what is the calculated power gain of an amplifier with a 5 mV signal at its input, and a 1 V signal at its output? _____ dB
19. The voltage gain of an amplifier can be adjusted from 1 (1:1) to 80 (80:1). What is its range of gain adjustment? _____ dB
20. If input and output impedances are matched, what amplifier voltage gain will yield a 26 dB power gain? Record your answer in gain factor (voltage ratio). ____:1

21. What is the power level of a 24 dBm signal? _____ mW
22. What is the power level of a -24 dBm signal? _____ μ W
23. What is the referenced power level of a 50 picowatt signal? _____ dBm
24. What is the referenced power level of a 50 watt signal? _____ dBm
25. A receiver is specified for a sensitivity of 20 dBf. What is this power level in femtowatts? _____ fW

CHAPTER 4 — Power/Frequency Applications

INTRODUCTION

In order to fully understand the quality of an electrical circuit or system, the level of a particular signal can be referenced to either time or frequency. An oscilloscope shows how the level of a signal changes over time, while a spectrum analyzer indicates how the level changes in regard to frequency.

A good way to study a signal is to graphically show how its level in dB or dBm varies over a particular range of frequencies. Figure 4-1 shows the frequency response of a typical good-quality audio amplifier.

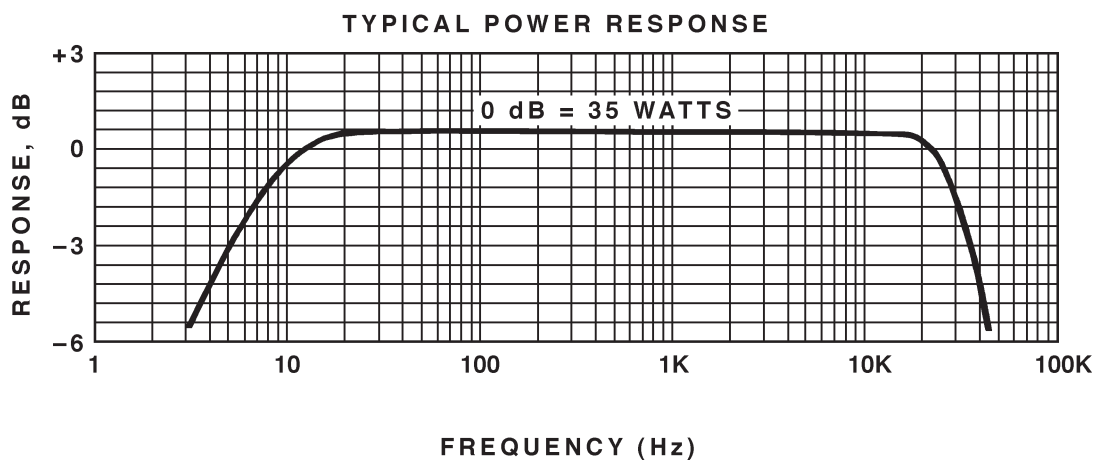


Figure 4-1

Typical response curve of an audio amplifier.

Note that the line is flat from 20 Hz to almost 20 kHz, and the level only drops to 3 dB at 5 Hz and 38 kHz, which are well above and below the range of normal human hearing.

Although the above example applies to acoustical signals, you will learn that you can also apply this technique to compare the strengths of other types of signals, such as radio broadcast stations.

Another important parameter of a signal is its signal-to-noise ratio, which indicates how much stronger the desired signal is than the noise level.

OBJECTIVES

When you complete this chapter, you will be able to:

- Define the concepts of frequency response (power vs. frequency) as it applies to telephone equipment.
- State the signal-to-noise ratios required for reliable communication with telephone and audio systems.
- Describe and dimension the characteristics of transmission lines and cables used for the efficient transfer of signal power.

ACOUSTICAL SIGNALS

Acoustical signals are sound pressure waves that cause our ear drums to vibrate. The range of human hearing is specified to be from 20 Hz to 20 kHz. Most young people can hear sounds within that range of frequencies. By mentioning frequency alone, however, we don't get a complete reading of hearing ability, or the relative intensity of different sounds. Sounds must be scaled in power levels, as well as frequency, to obtain a thorough indication of their properties. You are already familiar with electrical power. Now let's consider acoustical power.

Figure 4-2 shows the power levels of some representative acoustic signals in units of sound pressure level (dB SPL). Be aware that these acoustical levels, specified in decibels (dB) are not the same as the electrical power values that were introduced earlier. dB SPL is an absolute sound power level that is referenced to a very soft mechanical vibration, which is about as intense as a soft whisper. This 0 dB SPL reference is called the "threshold of hearing."

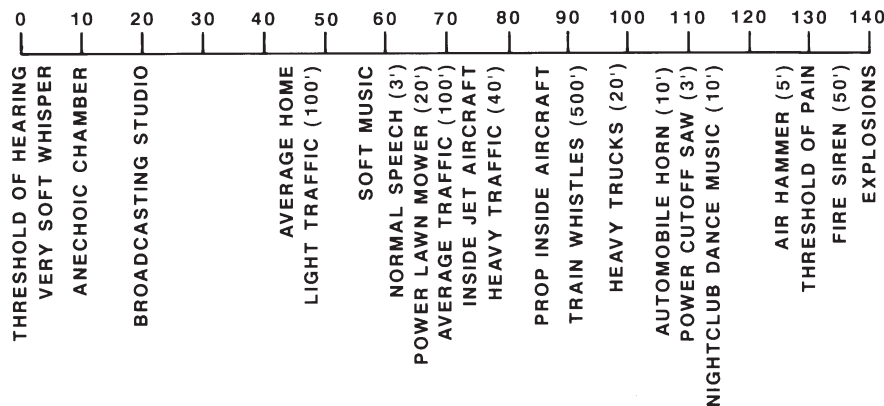


Figure 4-2

Power levels of acoustic signals.

Acoustical signal power is referenced power, scaled in dB SPL. Electrical signal power is also referenced power, but scaled in dBm.

While acoustical and electrical referenced signal power are distinct from each other, the power ratio-scaled in dB—for both acoustical and electrical signals are the same. A ten-fold increase in dB SPL is a 10 dB power ratio, just as a ten-fold increase in dBm is a 10 dB power ratio. Therefore, in Figure 4-2, the acoustic power level of the 90 dB train whistle is 100 times (20 dB) greater than the acoustic power of the 70 dB traffic sounds.

FREQUENCY RESPONSE OF SIGNALS

The most complete descriptions of signals can be displayed by combining their two dimensions of frequency and power into a single graph of frequency response. Figure 4-3 is such a graph. This figure shows how the human ear responds to sound signals at varying frequency and power levels. This single graph contains extensive information that would be difficult to convey with only single dimension scales.

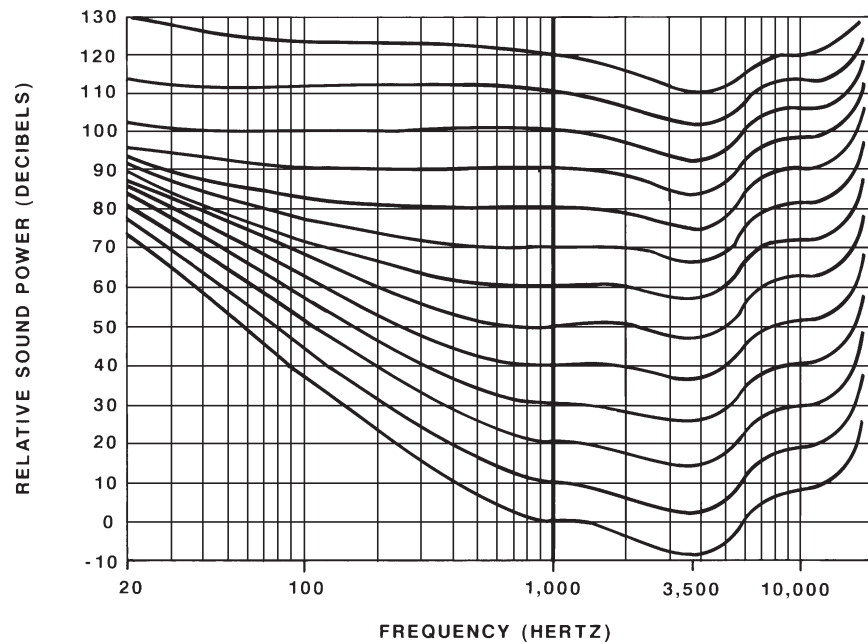


Figure 4-3

Response of human hearing.

The figure shows sound pressure levels that are referenced at 1000 Hertz. These signals are spaced at 10 dB power-level increments, from 0 dB SPL to 120 dB SPL. From the information on the graph you should see that the ear is 60 dB (1,000,000 times) less sensitive to 0 dB reference-level sounds at a frequency of 35 Hz, than it is to the same reference sounds at 1000 Hz. For high power-level sounds, the graph shows an equal sensitivity to 100 dB SPL signals from about 35 Hz to above 1000 Hz.

With just this short review of Figure 4-3 you should appreciate the great amount of information than can be conveyed by a typical graph of frequency response. It is for this reason that such graphs are used so extensively in the telecommunications field.

Frequency response graphs are used widely to describe electrical signals and the performance of electrical circuits and systems. Such graphs are often applied to the power distribution of signals over a range of frequencies. This type of graph is used to display the frequency locations and power levels of radio broadcast stations across the dial of a radio receiver in Figure 4-4. This figure shows the relative locations and power levels of received AM broadcast radio signals in the San Francisco, California (USA) area. Note that each signal's frequency is distributed along the X-axis of the graph. The signal power levels of the received stations are represented, in dBm, on the vertical Y-axis of the graph.

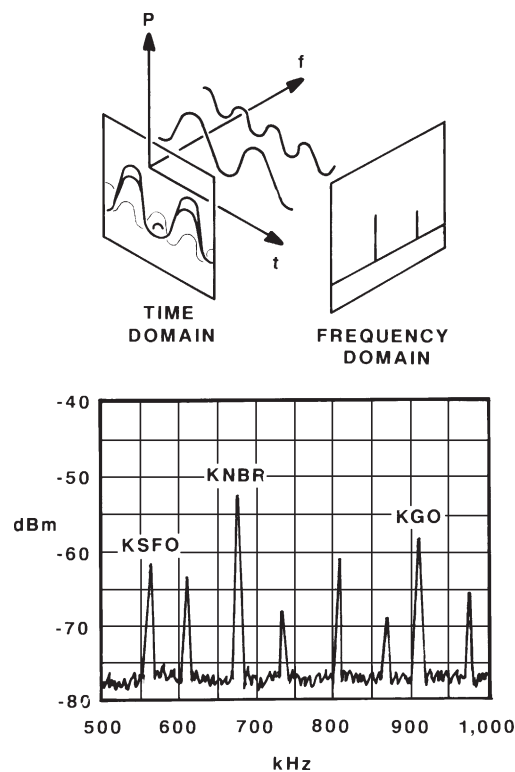


Figure 4-4

Frequency domain display of radio stations.

When signals are graphed in this way, they are said to be in the “frequency domain”. (In this context, the word “domain” is used to identify the controlled variable frequency—which is being graphed along the X-axis.) Direct measurements of power vs. frequency are often obtained with specialized test instruments that provide graphical displays. The spectrum analyzer is such an electronic test instrument. It measures and displays signal power and frequency in graphical form, like that shown in Figure 4-4. On the other hand, an oscilloscope measures and displays signals in the “time domain.” A comparison of frequency and time domains is shown in the top part of Figure 4-4.

Oscilloscopes measure and display signals in the time domain; spectrum analyzers measure and display signals in the frequency domain.

FREQUENCY RESPONSE OF CIRCUITS

Electrical and electronic components, circuits, and systems process signals. The signals are altered in some form as they pass through circuits. A perfect electronic circuit may either pass all signal frequencies applied to its input, or it may pass only a selected range of frequencies, depending upon its purpose. There are wide-band circuits, that pass an extended range of frequencies; and there are narrow-band circuits, that pass a limited range of frequencies. Each type of circuit is designed to perform a specific signal processing task. These circuits are usually evaluated in dB units of power gain or loss over a range of frequencies. Like the properties of signals, we also describe the performance of electronic components and circuits with time domain graphs. There is a major distinction between the frequency response graphs of components and circuits, and the frequency response graphs of signals. The graphs of signals show a number of distinct power levels at specific frequencies as in Figure 4-4; while graphs of components and circuits show a continuous variation of power level over a range of frequencies, as shown in Figure 4-5.

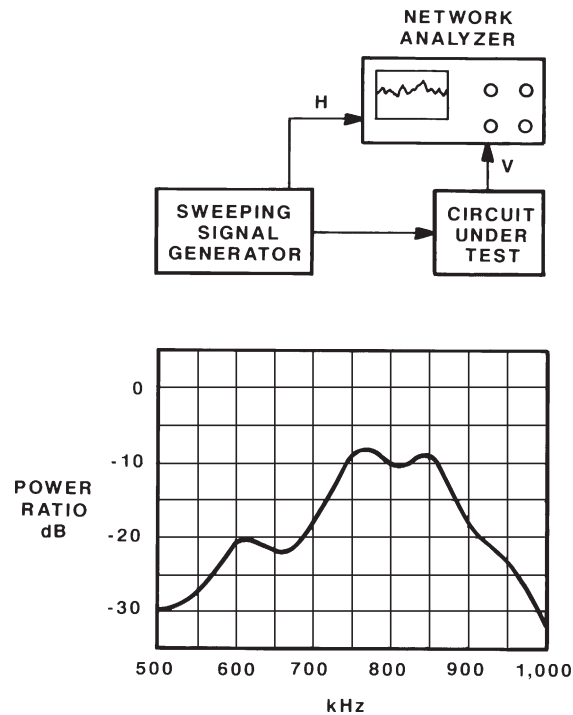


Figure 4-5

Frequency response display of a circuit.

There is also a difference in the test instruments used to measure and display the frequency response of components and circuits. The more popular of these instruments is the **network analyzer**. It is a frequency-domain CRT display instrument, like the spectrum analyzer, but the network analyzer requires a built-in or external signal generator to apply varying frequency signals to the components and circuits under test.

Spectrum analysis is the frequency-domain measurement of signals; while network analysis is the frequency-domain measurement of circuits.

Practical Examples

Figure 4-6A. shows the frequency response of a bandpass filter that has a minimum power loss over the range of frequencies from 2 kHz to 2.4 kHz; with increasing indications of power loss (scaled in dB) below 2 kHz, and above 2.4 kHz. The power loss for this filter, for a signal at 1 kHz, is shown to be 40 dB. In electronics jargon, this filter can be described as having “40 dB of rejection at 1 kHz.” The power level of a 1 kHz signal applied to this filter’s input will be reduced by 40 dB (1:10,000) as it passes through the component.

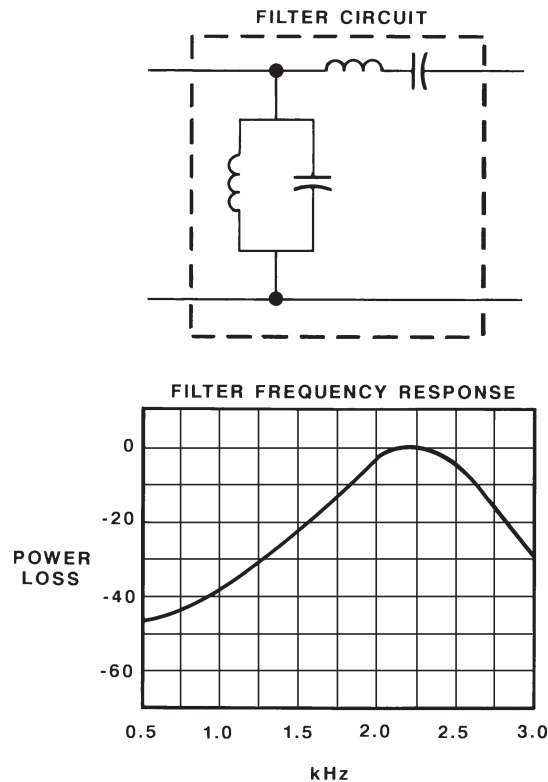


Figure 4-6

Frequency response of a bandpass filter.

Even transmission lines and cables, used for the transfer of telecommunications signals from one location to another, have frequency response characteristics that must be taken into account. A simple cable can be considered to be an equivalent RC electrical network, as shown in Figure 4-7A. Its typical frequency response is graphed with a solid line in Figure 4-7B. At the frequency where its shunt (parallel) capacitance is equal to the cable’s series resistance, the signal power loss through the cable is 3 dB. The engi-

neering term for this point is “corner frequency” or “first pole.” At frequencies above (higher than) this 3 dB point, the corner frequency, the signals will be further attenuated. The power ratio by which they will be reduced is 6 dB per octave, and 20 dB per decade. (An octave is a 2:1 frequency ratio, and a decade is a 10:1 frequency ratio.) With the 3 dB point at 10 kHz, the cable’s power loss can be predicted to be 9 dB (3 dB + 6 dB) at 20 kHz, and 23 dB (3 dB + 20 dB) at 100 kHz.

Above the corner frequency, the power in RC or RL networks varies at 6 dB/octave and 20 dB/decade.

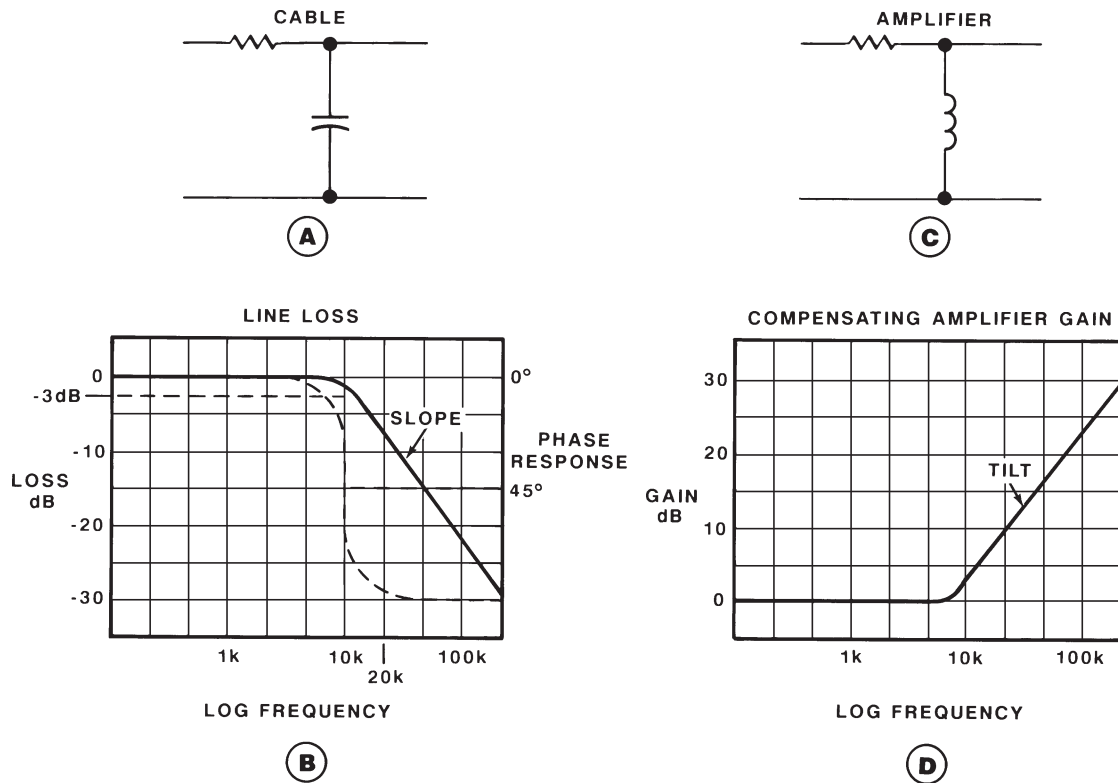


Figure 4-7

Cable and amplifier equivalent network and frequency response.

Very often cables are used at frequencies above their corner frequency, and they are said to have “slope.” That’s because they are being operated at frequencies where the graph of cable power vs. frequency slopes downward.

Amplifiers are often placed in longer distance transmission paths where such cables are used. This is represented by the equivalent RL electrical network in Figure 4-7C. These amplifiers compensate for the cable’s losses, as they have a designed-in frequency response opposite to that of the cable loss. The amplifiers have increasingly higher gain—6 dB per octave—at increasingly higher frequencies. Such amplifiers are said to have “tilt,” because their graph of frequency response tilts upward, as shown in Figure 4-7D.

Phase Response

The right-side Y-axis of the cable network graph of Figure 4-7B is scaled in units of phase shift, corresponding to the dashed line plotted in the graph. The frequency response graph also displays the corresponding phase-response of the network over the same range of frequencies as signal loss is shown.

When you studied basic electrical network theory and AC electrical fundamentals in your earlier courses, you probably learned that a phase shift of 45° occurs when the reactance and resistance of a network are equal. This phase shift point is also the point where the signal power level drops 3 dB the corner frequency. At frequencies above this point, the phase difference between signal voltage and current shifts even further; resulting in increased signal loss (remember the power factor).

The concepts of frequency response, such as described with the cable losses and compensating amplifier operation of Figure 4-7, are used in many audio, telephone, and telecommunications systems. One example is the frequency compensation of audio tape, called “equalization.” Another example is the frequency compensation of the audio in FM broadcast radio, and in telephone multiplex systems, called “pre-emphasis” and “de-emphasis.” Your thorough understanding of frequency response fundamentals here, will make your later studies of telecommunications systems, much easier.

SIGNAL-TO-NOISE RATIOS

Recall that noise signals are random in frequency and equally distributed in power level across the range of frequencies that pass through telecommunications circuits. Noise that has slope, lower power at higher frequencies, is called “pink noise;” while noise that has a uniform power level across a range of frequencies, is called “white noise.” Noise occurs as a result of the random movement of electrons in circuits (and in air). This type of noise occurs as a result of atmospheric and electrical disturbances such as the discharge of lightning and static charges, and as a result of environmental effects of sparking and arcing.

Any random movement of electrons that doesn't contribute to the generation and transfer of information is considered to be noise.

Telecommunications signals must operate in the presence of noise. Noise does not degrade communicated information if the noise power level is low when compared to the power level of the information signals. When the noise power level approaches the level of the information signal, the information becomes degraded, and may even become useless. The object is to keep the power levels of information signals as far above the noise levels as possible. The measure of the power ratio between these two levels is called the “signal-to-noise” ratio. Signal-to-noise ratio (abbreviated S/N) is measured in dB units.

For reliable communications, you must keep the power levels of information signals as far above the noise levels as possible.

Certain communications services require a minimum signal-to-noise ratio for reliable communication, and they attempt to maintain some higher S/N for “quality” service. For example, voice telephone and broadcast radio services attempt to provide at least 60 dB S/N. You can completely understand telephone conversations with 20 dB S/N, but the background hiss will be very annoying. With less than 10 dB S/N, you won't understand all of the words.

You are not likely to enjoy listening to music with S/N values less than 30 dB, and you will appreciate higher S/N values in music recordings. The current state-of-the-art musical recording medium, digital disks (CDs), maintain S/N values in excess of 90 dB!

In video and television, noise shows up as snow in the picture; and in satellite TVRO receivers, noise shows up as sparkle in the picture. A minimum of 40 dB S/N is desired in video and television images.

Figure 4-8 illustrates two methods used to observe signal-to-noise ratios. On the left is an oscilloscope time domain display. On the right is a spectrum analyzer frequency domain display.

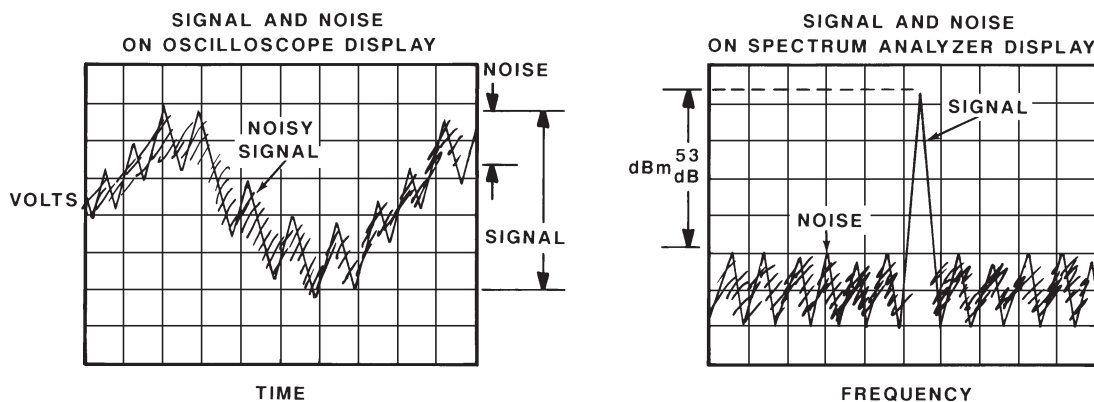


Figure 4-8
Signal-to-noise ratios.

When practical, signal power levels are adjusted for a desired signal to-noise ratio. For example, if a measurement of noise power is indicated at 100 pW (−70 dBm) in a wired telephone system, a 60 dB S/N will result with a signal power level 60 dB above noise, at 100 μW (−10 dBm).

$$-70 \text{ dBm} + 60 \text{ dB} = -10 \text{ dBm}$$

Radio receiver sensitivity specifications are also based upon signal-to noise ratios. For example, an FM broadcast stereo receiver has a listed sensitivity of 17 dBf (50 fW). This specification is based upon a 30 dB S/N. The noise power is 30 dB below the signal level of 17 dBf, a power level of −13 dBf (.05 fW).

$$17 \text{ dBf} - 30 \text{ dB} = -13 \text{ dBf}$$

From these examples you may conclude that the noise levels of received VHF FM broadcast radio are far lower in power than those within wired telephone systems.

SIGNAL POWER TRANSFER

One of the major purposes of telecommunications systems is to transfer information from one location to another using electrical signals. This function is accomplished through wires and cables, and through the air or space with radio antennas. The goal is to achieve this transfer of information with a minimum of signal loss, while maintaining high signal-to-noise ratios. Efficient transfer of electrical signals with minimum power loss requires a uniform path for transmission. In electrical terms, the requirement is for uniform impedances at all locations within the electrical circuits through which signals pass.

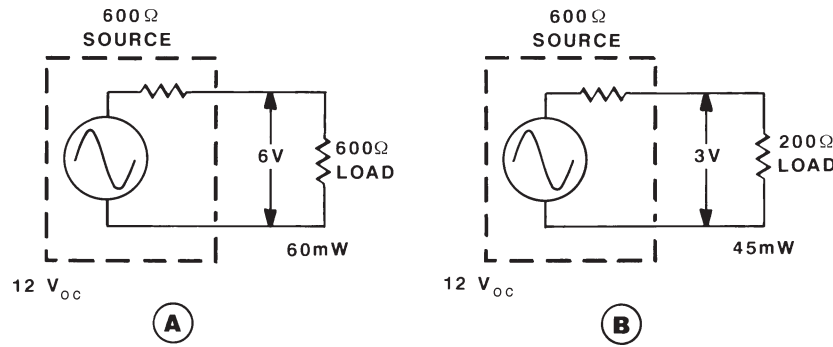
You may recall the Power Transfer Theorem from your earlier studies of electrical fundamentals. The theorem states that maximum power transfer occurs between a source of power and a circuit load, when the load resistance matches the internal resistance of the power source. Transmission circuit impedances must be set or adjusted to match the signal source impedances in telecommunications systems.

In practice, specific impedance values have been standardized to make impedance matching for power transfer a practical reality. The most popular of these impedances are 600 ohms for audio and telephone circuits, 300 ohms for television broadcast reception, 75 ohms for the interconnection of television, video, and computer equipment, and 50 ohms for RF and microwave systems.

Maximum power transfer occurs between a signal source and a circuit load, when the load resistance matches the internal resistance of the source.

Mismatch Loss

The power loss that occurs, due to impedance mismatches in transmission circuits, is compared to the ideal of maximum power transfer. The ratio of the two signal levels, mismatched vs. matched, is called “mismatch loss.” Mismatch loss is designated (and measured) in dB units. Mismatch loss is directly proportional to the ratio of impedance mismatch, and can be quickly calculated as such. However, let’s first review some basic electrical network theory to convey the concept of mismatch.

**Figure 4-9**

Power in a mismatched load.

Let's compare the power delivered to a 200-ohm load from a 600-ohm signal source, using the simple equivalent circuits of Figure 4-9. With a matched impedance, and a load power level of 60 mW with 6 volts applied across a 600 ohm load as shown in Figure 4-9A, the open-circuit voltage is 12 volts. With the load resistance at 200 ohms as shown in Figure 4-9B, the load voltage will drop to 3 volts; resulting in a power level of 45 mW. Comparing the mismatched power to the matched power yields 45 mW versus 60 mW, or a power level that is 75% of maximum. In logarithmic units, 75% is a power loss of 1.25 dB. The step-by-step procedures used to obtain these numbers are summarized as follows:

1. The power (P) at the load is calculated from the information given as 6 volts across 600 ohms:

$$P = \frac{V^2}{R} = \frac{(6V)^2}{600 \Omega} = 60 \text{ mW}$$

2. The open-circuit voltage (V_{OC}) of this 600-ohm signal source is calculated as twice the load voltage (V_L) in this matched circuit:

$$V_{OC} = 2 V_L(\text{matched}) = 2 \times 6V = 12V$$

3. The voltage (V_L) across the 200-ohm load (R_L) is calculated using the voltage-divider formula (R_S is the 600-ohm source resistance):

$$V_L = V_{OC} \left[\frac{R_L}{(R_S + R_L)} \right]$$

$$V_L = 12V \left[\frac{200 \Omega}{(600 \Omega + 200 \Omega)} \right] = 3 V$$

4. The power of the 200-ohm load is calculated as:

$$P = \frac{V^2}{R_L}$$

$$P = \frac{(3V)^2}{200 \Omega}$$

$$P = 45 \text{ mW}$$

5. Finally the mismatch loss (in dB) is calculated from the power ratios:

$$\text{dB} = 10 \log_{10} \left(\frac{P_2}{P_1} \right)$$

$$\text{dB} = 10 \log_{10} \left(\frac{45 \text{ mW}}{60 \text{ mW}} \right)$$

$$\text{dB} = -1.25$$

Once you understand how these calculations result in a given value of mismatch loss, you may appreciate a far simpler approach. This calculation procedure uses a single equation based upon impedance ratios. (In the following equation, Y is the ratio of the two impedances.)

$$Y = \frac{R_S}{R_L} = \frac{600 \Omega}{200 \Omega} = 3$$

$$\text{dB} = 10 \log_{10} \left[1 - \left\{ \frac{(Y-1)}{(Y+1)} \right\}^2 \right]$$

$$\text{dB} = 10 \log_{10} \left[1 - \left\{ \frac{(3-1)}{(3+1)} \right\}^2 \right] = -1.25 \text{ dB}$$

Transmission Lines

Transmission lines and cables are used to transfer signal power from one location to another in telecommunications systems. These lines serve to transfer the signal power from the signal source to the circuit load. To carry this signal power without mismatch loss, the lines must themselves have an impedance value that is matched to the signal source. This impedance is called “characteristic impedance,” designated with the abbreviation Z_0 . Characteristic impedance is a distinct kind of impedance that allows signals to pass along transmission circuits with a minimum of power loss from either impedance mismatch or attenuation.

Characteristic impedance (Z_0) is the electrical property of cables that provides for impedance matching and maximum power transfer to a load.

You should recall the concept of power factor, where true power is degraded by the extent that signal voltage and current are out of phase. In the extreme, signal voltage and current are 90° out of phase, and there is zero true power dissipation. This is the condition under which signals pass along transmission lines and cables. The ideal cables are completely reactive, with 90° phase angles, so there is no power dissipated by the signals passing along them. When the purely reactive characteristic impedance ($Z_0 = V/I$) of the cables match the internal impedance of the signal source, there will be an optimum transfer of power to the cables. The cables will, in turn, carry this signal power to the circuit load, maintaining the correct voltage-to-current ratio to drive a matched load impedance.

To serve the purposes of optimum power transfer, cables and their related connector families are designed to have minimum resistance and maximum reactance. They are available to match the impedances of standardized telecommunications circuits and systems; including those more popular values noted earlier: 600 ohms, 300 ohms, 75 ohms, and 50 ohms. It is very important to understand the function provided by transmission lines and cables, without which the best of information signals would not go very far.

SUMMARY

Decibels are not only used to scale electrical signal power ratios, they are also used to scale acoustical power ratios. Acoustical signals are the sound pressure waves that make our eardrums vibrate. Sound pressure levels are designated as dB SPL. dB SPL is an absolute sound power level referenced to a very soft mechanical vibration. The 0 dB SPL reference is called the “threshold of hearing.”

The most complete description of a signal is obtained by displaying its frequency response. Frequency response is a single graphed display of the frequency and power dimensions of a signal.

Noise is an important factor affecting telecommunications circuits. Noise is always present, but as long as the noise power level is low when compared to the signal power level, it does not degrade the communicated information. For this reason, a minimum signal-to-noise ratio must be maintained to provide reliable, quality communications services.

SELF-TEST REVIEW

1. Sound pressure levels are measured in:
 - A. dB.
 - B. dBm.
 - C. volts.
 - D. watts.
2. Signal gain and loss are measured in:
 - A. dB.
 - B. dBm.
 - C. volts.
 - D. watts.
3. At which approximate frequency is the human ear most sensitive to soft (quiet) sound signals?
 - A. 100 Hz
 - B. 400 Hz
 - C. 1000 Hz
 - D. 4000 Hz
4. What information is not available from measurements made using the spectrum analyzer?
 - A. Frequency.
 - B. Time.
 - C. Voltage.
 - D. Power.
5. What instrument provides a continuous graph of the frequency response of electronic circuits?
 - A. Oscilloscope.
 - B. Frequency meter.
 - C. Spectrum analyzer.
 - D. Network analyzer.

6. What instrument provides graphical display of individual signal power levels over a range of frequencies?
 - A. Oscilloscope.
 - B. Frequency meter.
 - C. Spectrum analyzer.
 - D. Network analyzer.
7. A 10 mW signal is applied to the input of a filter with 20 dB rejection at the frequency of that signal. What is the signal power level at the filter's output?
 - A. 10 mW
 - B. 1 mW
 - C. 0.1 mW
 - D. 0.01 mW
8. A transmission circuit, with RC frequency response, has a corner frequency of 2 kHz. What is the attenuation of the circuit for a signal frequency of 8 kHz?
 - A. 6 dB
 - B. 9 dB
 - C. 12 dB
 - D. 15 dB
9. A transmission circuit, with RC frequency response, has a "first pole" frequency of 2 kHz. What is the attenuation of the circuit for a signal frequency of 20 kHz?
 - A. 20 dB
 - B. 23 dB
 - C. 26 dB
 - D. 30 dB
10. What is the reactance at the corner frequency of an RC response 600-ohm transmission circuit?
 - A. 300 Ω
 - B. 600 Ω
 - C. 900 Ω
 - D. 1200 Ω

11. What is the phase shift at the corner frequency of an RC response transmission circuit?
 - A. 30°
 - B. 45°
 - C. 60°
 - D. 90°
12. What signal power is required for 40 dB S/N, and a noise power level of 200 nanowatts?
 - A. 20 μ W
 - B. 200 μ W
 - C. 2 mW
 - D. 20 mW
13. What is the noise power of a receiver with a sensitivity of 22 dBf, and a S/N of 24 dB?
 - A. -2 dBf
 - B. +2 dBf
 - C. 24 dBf
 - D. 46 dBf
14. What value of cable and load impedance is required for maximum power transfer when interconnecting computer equipment?
 - A. 50 Ω
 - B. 75 Ω
 - C. 300 Ω
 - D. 600 Ω
15. What approximate value of mismatch loss will occur when a 300-ohm transmission line is used in a circuit designed for 600-ohm lines?
 - A. 0.5 dB
 - B. 1 dB
 - C. 1.5 dB
 - D. 2 dB
16. A transmission circuit, with RC frequency response, has a "first pole" frequency of 2 kHz. What is the attenuation of the circuit for a signal at a frequency of 40 kHz? _____ dB

17. What is the S/N (in dB) with a 2 mW signal level, and a 100 pW noise level? _____ dB
18. What signal level is required for a 30 dB S/N, if the noise level is at 100 pW? _____ nW _____ μ W
19. What value of mismatch loss will result if a 75-ohm cable is used in a 600-ohm transmission circuit? _____ dB

Section 2

**WAVEFORMS
AND
HARMONICS**

CHAPTER 5 — Frequencies Within Signal Waveforms

INTRODUCTION

In Chapters 1 through 4 we introduced the fundamentals of signal power and frequency. These are the basic dimensions of signals used in telecommunications systems. Beginning with this chapter we will describe another important aspect of signals that involves power and frequency: waveform and harmonic content. Signals with waveforms other than sine waves have more than one frequency of operation. They include a number of sine-wave frequencies simultaneously carried within their waveform. You will now learn how to identify the frequencies and power levels of these many signals within a variety of waveforms.



OBJECTIVES

When you complete this chapter, you will be able to:

- Define the fundamental and harmonic frequencies of electrical signals.
- Identify common signal waveforms, and how they can be created by combining multiple frequency sine waves.

THE IDEAL SINE WAVE

The sine waveform is obtained by graphing the rotation of any uniformly pivoting body. You may recall (from Chapter 1) that signal phase and wavelength were introduced using the example of a bicycle moving along a road. The bicycle's speed is uniform and continuous, because the rotation of the wheels are at a uniform velocity. A graph of the tire's rotation will produce a perfect sine wave. The name "sine" is given to this waveform that is graphed in a shape that agrees with the calculation of trigonometric values based upon a perfect circle. The sine wave is an *ideal* waveform, and the standard against which other wave shapes are compared.

Electrical power is most often generated by uniformly rotating machines called "alternators." The output voltages of these machines are proportional to their rotation, just as the mechanical force of a wheel varies in proportion to its rotation. The outputs of such generators are sine waves. The same sine waveform can also be obtained from circulating currents, within the resonant networks of electronic circuits.

Sine waveforms are analyzed mathematically to determine their changing voltage levels with respect to their phase (rotation angle). The voltage levels, and the currents that result from applying them to a circuit, are called "instantaneous" voltage and current. The instantaneous value of voltage (V_i) or current (I_i) at any point on the sine waveform can be calculated as:

$$V_i = V_{\text{peak}} \sin \angle \theta \text{ and } I_i = I_{\text{peak}} \sin \angle \theta$$

The peak voltage (V_{peak}) or peak current (I_{peak}) is the maximum level to which the sine wave rises; and $\angle \theta$ the phase angle of the waveform at the point of calculation. Using this formula, the instantaneous voltage of a 2-volt peak sine wave at the 30° phase point of the waveform is calculated as follows:

$$V_i = V_{\text{peak}} \angle \theta = 2 \times \sin 30^\circ = 1 \text{ V}$$

Figure 5-1 shows the magnitudes of a sine wave at 30 degree increments over one cycle of its phase shift. (You should recall that the magnitude of the sine wave at 45° represents its effective rms value of $0.707 \times$ its peak value at 90° .)

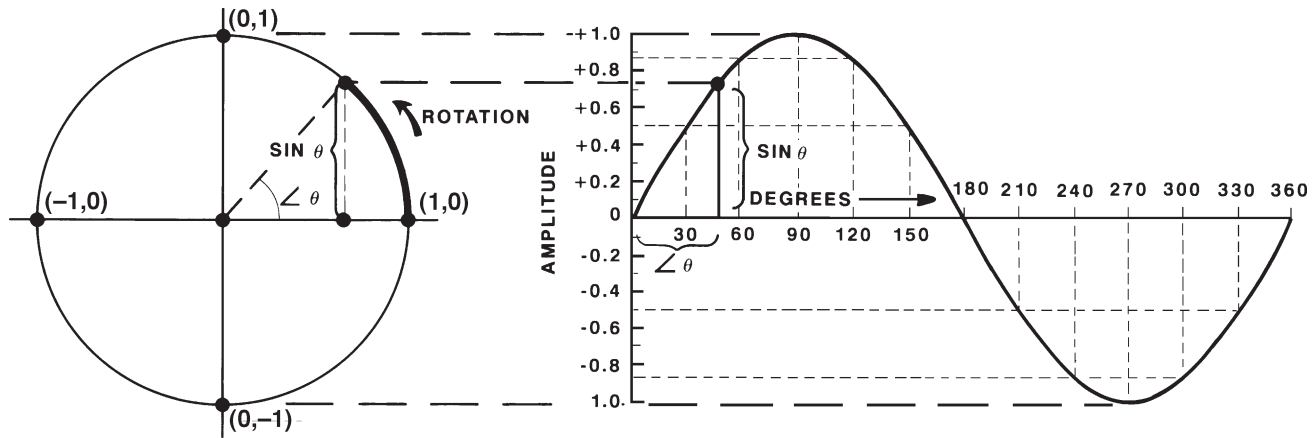


Figure 5-1

Generating the sine wave from circular rotation.

As you study Figure 5-1, keep in mind that the sine wave is distinctive, and it is the waveshape of *all* continuous signal frequencies. Waves with shapes other than sine waves are made up of a number of combined individual sine waves of different frequencies. A perfect sine wave contains only *one* single frequency.

All waveforms are comprised of sine waves. Non-sine waveforms are a result of individual sine-waves of different frequencies.

Non-Sine Waveforms

Non-sine waveforms can be created by combining a lower frequency sine wave signal with other sine wave signals that are at frequency multiples of the lower frequency. The lowest frequency signal of the group is always called the “fundamental,” while the higher frequency signals are called “harmonics.” The harmonic signals are individually identified by their multiple factor. The second harmonic is $2\times$ the fundamental’s frequency, the third harmonic is $3\times$ the fundamental’s frequency, and so on.

The lowest frequency signal of any waveform is the “fundamental.” “Harmonics” are higher frequency signals numbered as multiples of the fundamental.

All waveforms other than perfect sine waves have harmonics. Non-sine waveforms are, in fact, identified as “complex,” because they are comprised of more than one frequency. Virtually all signal waveforms that are used to transfer voice and other information in telecommunications systems are complex.

The Effects Of Combining Signals

In the process of combining signals, their instantaneous voltages will sum (add), and the result is a waveform shaped as a summation of the signals. As an example, we can make a series of calculations to show how the combination of the fundamental and third harmonic frequencies will affect each other. For this example, we will assume the peak amplitude of the fundamental is 3 volts, and the peak amplitude of the third harmonic is 1 volt. (The frequency of the third harmonic is $3\times$ the fundamental, so it will have $3\times$ the phase shift of the fundamental: 30° for the fundamental is 90° for the third harmonic.)

Procedure:

At 0° , both signal waveforms are at 0 V

$$V_{\text{ifund}} = V_{\text{peak}} \sin \angle \theta = 3 \sin 0^\circ = 0 \text{ V}$$

$$V_{\text{ihar3}} = V_{\text{peak}} \sin \angle \theta = 1 \sin 0^\circ = 0 \text{ V}$$

$$V_{\text{itotal}} = V_{\text{ifund}} + V_{\text{ihar}} = 0 + 0 = 0 \text{ V}$$

At 30° , the complex waveform, the result of summing the fundamental and third harmonic waveforms, rises to the higher value of 2.5 volts, compared to the fundamental sine wave at 1.5 volts:

$$V_{\text{ifund}} = V_{\text{peak}} \sin \angle \theta = 3 \sin 30^\circ = 1.5 \text{ V}$$

$$V_{\text{ihar3}} = V_{\text{peak}} \sin \angle \theta = 1 \sin 90^\circ = 1 \text{ V}$$

$$V_{\text{itotal}} = V_{\text{ifund}} + V_{\text{ihar}} = 1.5 + 1 = 2.5 \text{ V}$$

At 60° , the complex waveform reaches a level equal to the sine wave at 2.6 volts:

$$V_{\text{ifund}} = V_{\text{peak}} \sin \angle \theta = 3 \sin 60^\circ = 2.6 \text{ V}$$

$$V_{\text{ihar3}} = V_{\text{peak}} \sin \angle \theta = 1 \sin 180^\circ = 0 \text{ V}$$

$$V_{\text{itotal}} = V_{\text{ifund}} + V_{\text{ihar}} = 2.6 + 0 = 2.6 \text{ V}$$

At 90° , the complex waveform drops to 2 volts, which is far below the peak value of the sine wave at 3 volts:

$$V_{\text{ifund}} = V_{\text{peak}} \sin \angle \theta = 3 \sin 90^\circ = 3 \text{ V}$$

$$V_{\text{ihar3}} = V_{\text{peak}} \sin \angle \theta = 1 \sin 270^\circ = -1 \text{ V}$$

$$V_{\text{itotal}} = V_{\text{ifund}} + V_{\text{ihar}} = 3 + -1 = 2 \text{ V}$$

If you apply these same calculations at finer increments, you will obtain the waveshape shown in Figure 5-2A. Figure 5-2B shows what happens to the complex waveform if you add the fifth harmonic. Note the waveform is approaching the shape of a square wave. Square waves are comprised of odd-numbered harmonics. The fundamental frequency is summed with harmonics that are multiples of three, five, seven, and nine times the lower frequency. As an example, let's assume that the fundamental frequency (f_{fund}) of a signal is 400 Hz. The harmonics shown include the third, fifth, seventh and ninth. The harmonic frequencies (f_{har}) will be:

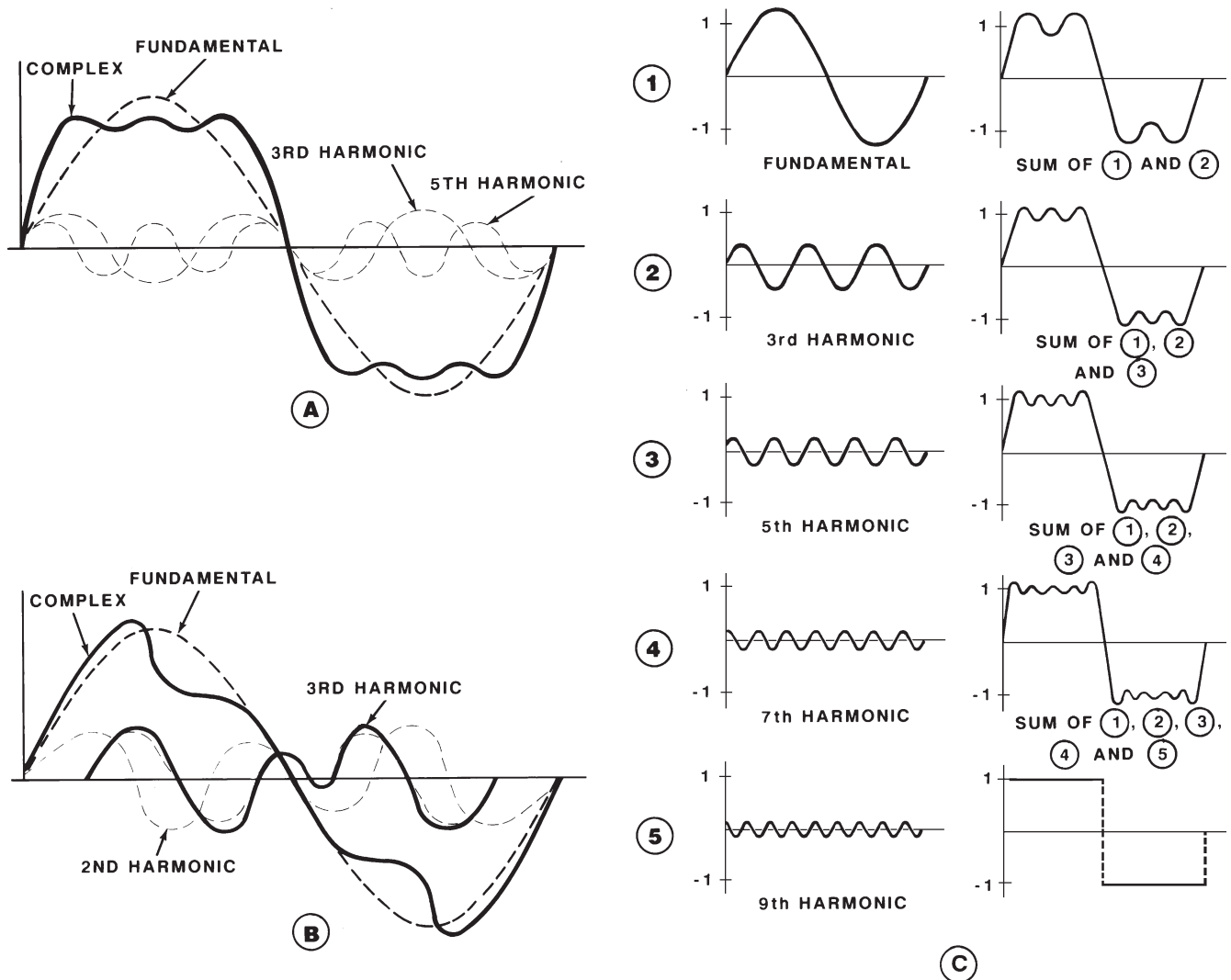
$$f_{\text{har}} = f_{\text{fund}} \times \text{harmonic number}$$

$$f_{\text{har3}} = 400 \times 3 = 1200 \text{ Hz}$$

$$f_{\text{har5}} = 400 \times 5 = 2000 \text{ Hz}$$

$$f_{\text{har7}} = 400 \times 7 = 2800 \text{ Hz}$$

$$f_{\text{har9}} = 400 \times 9 = 3600 \text{ Hz}$$

**Figure 5-2**

Non-sine waveforms from multiple frequencies.

The spectrum of this signal's waveform will include frequencies from 400 Hz through 3600 Hz; with the harmonics spaced at 800 Hz increments above the *fundamental* of 400 Hz.

Sawtooth waveforms are comprised of all (both odd and even) harmonic frequencies. If you assume the fundamental frequency of a sawtooth signal to be 400 Hz, its harmonics will include frequencies of 800, 1200, 1600, 2000, 2400, 2800, 3200, and 3600 Hz. The spectrum of the sawtooth signal's waveform will include frequencies from 400 Hz through 3600 Hz; with its *harmonics* spaced at 400 Hz increments above the *fundamental* of 400 Hz.

WAVEFORMS CONTAINING MULTIPLE FREQUENCIES

The *exact* waveshape that results from combining a fundamental frequency signal with its harmonics, will depend upon *both* the numbers *and* power levels of the harmonic signals. Figure 5-3 shows graphic displays of harmonics and their respective power levels for some common waveforms.

In studying Figure 5-3, you should observe that the square waveform contains odd-numbered harmonics (3, 5, 7, etc.), at decreasing power levels. By contrast, the pulse waveform includes all harmonics (2, 3, 4, 5, 6, etc.), with some higher power levels in the harmonics than in the fundamental. The triangle, ramp, and rectified sine waveforms are also distinctive.

There is no limit to the number of harmonic combinations and the particular waveforms that they will produce. Each waveform will have its own distinctive harmonic structure, which is its *signature*. While it is impossible to memorize the harmonic structure of all waveforms, you should be generally familiar with some common waveforms. The general harmonic structure of the more common waveforms are similar to those shown in Figure 5-3.

A signal's waveshape depends upon its distinctive harmonic structure, the number and power levels of its harmonics.

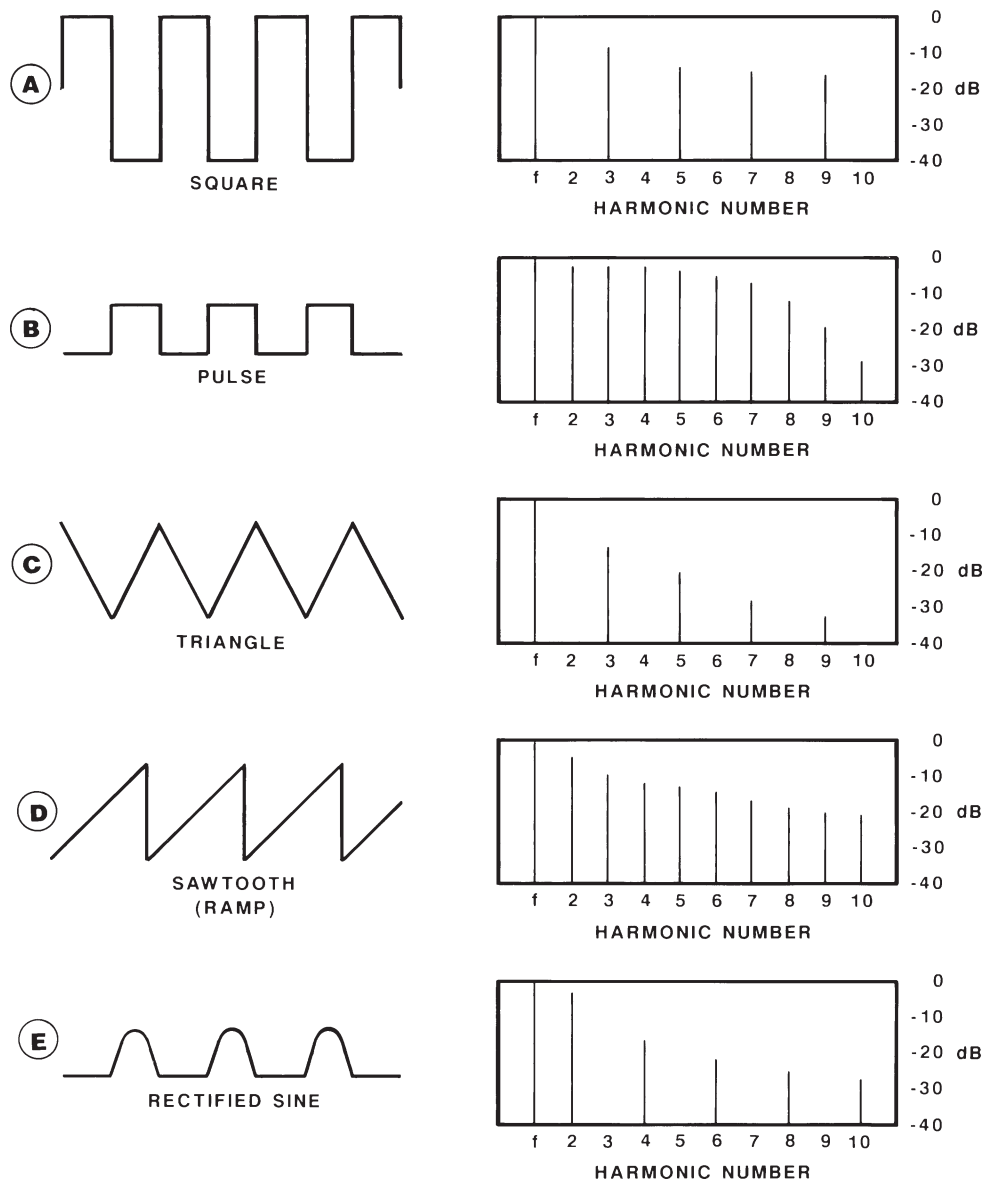


Figure 5-3
Harmonic spectra of non-sine waveforms.

It is also often necessary to determine the exact harmonic content of a specific waveform. For such requirements, it is practical to measure the waveform directly using a spectrum analyzer. The spectrum analyzer will display the instantaneous power and frequency of *any* signal waveform and its harmonics. The analyzer eliminates the need to separately measure *each* of the individual harmonic signals. Figure 5-4 shows a typical signal displayed both on the spectrum analyzer and the oscilloscope.

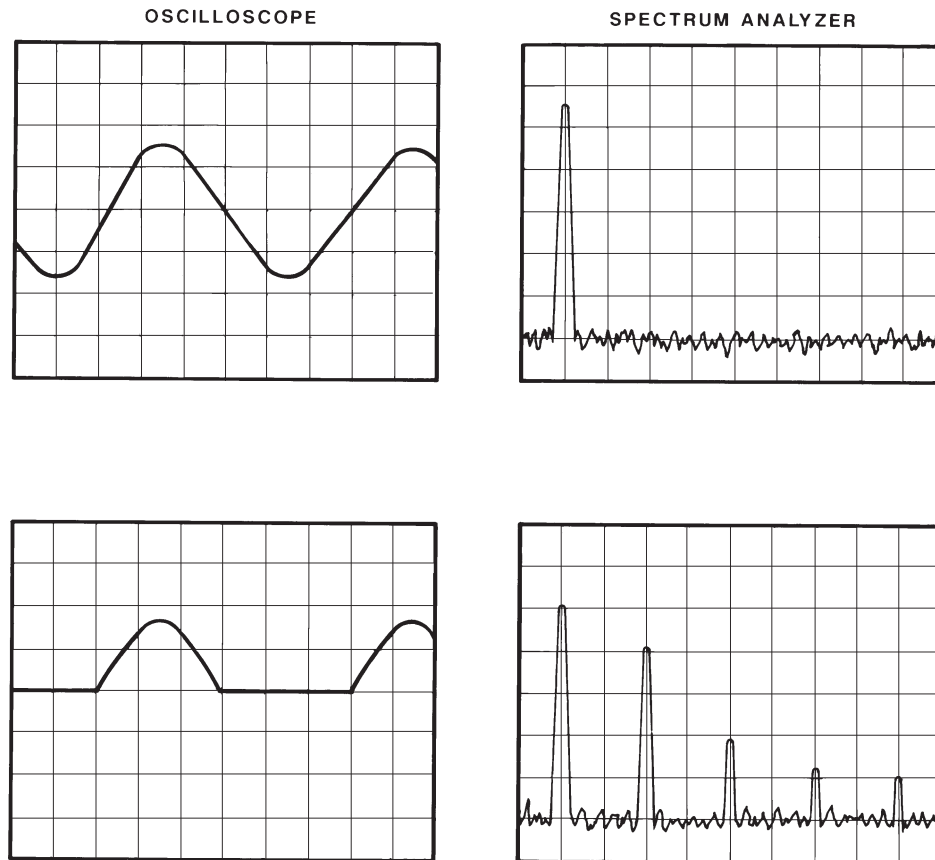


Figure 5-4

Oscilloscope and spectrum analyzer displays.

Creating Waveforms

The method of combining sine-wave signals to produce specific waveforms is impractical. We have used this approach *only* to verify that all waveforms are comprised of certain combinations of sine waves. Actually, almost any wave shape can be created electronically, and much more practically, than with multiple frequency signal sources. There are many IC devices and simple circuits that can produce all of the common waveforms of Figure 5-3.

Square waves and pulses are usually produced using electronic circuits that switch on and off at some fundamental frequency. Triangle and ramp (sawtooth) waveforms can be generated from similar circuits that use capacitors that charge and discharge as the circuits switch on and off. Sine waves can be produced using circuits that switch on and off across a resonant LC network; and rectified sine waves can be obtained by passing the sine wave through a simple diode. Figure 5-5 shows some simplified circuits that produce these common waveforms.

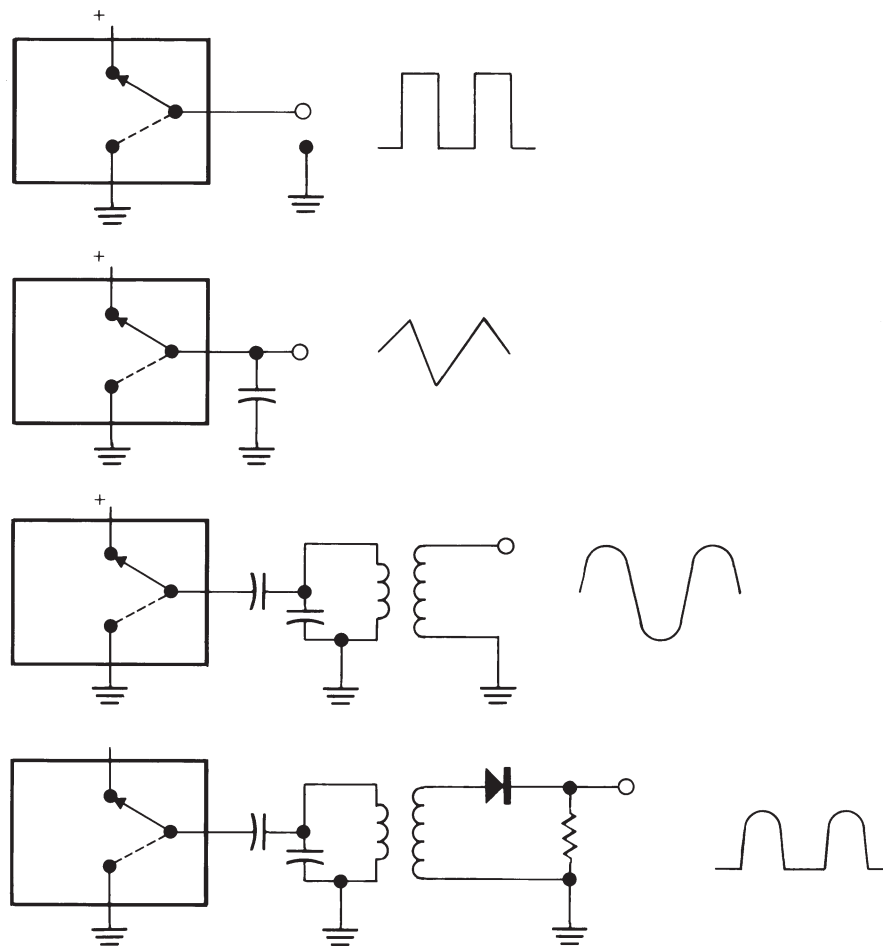


Figure 5-5
Generating non-sine waveforms.

Communications Signals

The common waveforms, such as those of Figure 5-5, are easily generated, and their harmonic structure is generally predictable. Unfortunately the signals that we process through audio, telephone, or video equipment seldom have waveforms that are common or predictable. The human voice is a good example of a signal source with a variety of waveforms. When you speak, even a slight change in the pitch (frequency) of your voice will change its waveform. And overall, it is the unique harmonic structure of your voice that makes your speech distinct from that of others.

Voice signals processed through a telephone transmitter and receiver provide an excellent example of circuit effects upon complex waveforms. The ideal sound system in a recording studio will reproduce every waveform of your speech accurately. However, different telephone circuits will affect the waveform of your speech, and the tone of your voice will sound different. The telecommunications industry has made considerable effort to design equipment that will accurately process voice waveforms, but you can still hear a lot of variation from one telephone set to another.

SUMMARY

In this chapter, we discussed sine waves and non-sine waves. Non-sine waves are also referred to as complex waves. The fundamental is always the lowest frequency in a complex waveform, while the higher frequency multiples of the fundamental are called harmonics.

When signals are combined, their instantaneous voltages will add, resulting in a waveform shaped as a summation of the signals.

SELF-TEST REVIEW

1. At what angle of rotation is the maximum voltage in a sine waveform?
 - A. 0 deg
 - B. 30 deg
 - C. 60 deg
 - D. 90 deg

2. Two sine waves of equal voltage level, with one at a frequency three times higher than the other, are combined. At what angle of rotation is the maximum voltage in the waveform that results from this combination?
 - A. 0 deg
 - B. 30 deg
 - C. 60 deg
 - D. 90 deg

3. A signal's fundamental is at 400 Hz, at what frequency is its third harmonic?
 - A. 400 Hz
 - B. 800 Hz
 - C. 1200 Hz
 - D. 1600 Hz

4. Which waveform is not identified as "complex?"
 - A. Sine
 - B. Triangle
 - C. Square
 - D. Pulse

5. Which waveform has predominantly odd-numbered harmonics?
 - A. Sine
 - B. Ramp
 - C. Square
 - D. Pulse

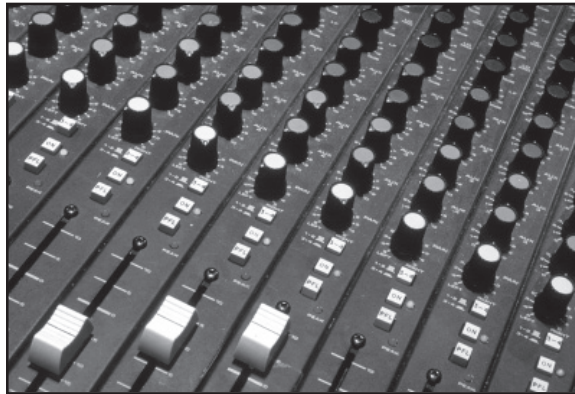
6. What is the spectral width of a 400 Hz sawtooth waveform with the first four significant harmonics above its fundamental?
 - A. 1600 Hz
 - B. 2000 Hz
 - C. 2400 Hz
 - D. 2800 Hz
7. What is the fundamental frequency of a signal with its fourth harmonic at 2000 Hz? _____ Hz
8. Which signal property requires measurement with a spectrum analyzer, rather than an oscilloscope?
 - A. Frequency
 - B. Power
 - C. Waveform
 - D. Harmonic distortion
9. Which waveform results from electronic circuits that switch on and off across a resonant LC network?
 - A. Sine
 - B. Triangle
 - C. Square
 - D. Pulse
10. Which harmonic number has the highest power level in a rectified sine waveform?
 - A. second.
 - B. third.
 - C. fourth.
 - D. fifth.
11. A sinewave signal has a peak voltage of 5 volts. What are the instantaneous voltages on the waveform at phase angles of:
 - A. 0 deg _____
 - B. 30 deg _____
 - C. 45 deg _____
 - D. 60 deg _____
 - E. 90 deg _____

12. A sinewave signal with a peak voltage of 5 volts, is combined with another signal with a peak voltage of 2 volts. The 2-volt signal is operating at three times the frequency of the 5-volt signal, and both signals are in phase at 0° of their phase. The resulting combined signal waveform will not be a sine wave. What are the instantaneous voltages of the combined signal at:
- A. 0 deg _____
 - B. 30 deg _____
 - C. 45 deg _____
 - D. 60 deg _____
 - E. 90 deg _____
13. What is the power ratio, in dB below the fundamental, for the third harmonic of the following waveforms? (Refer to Figure 5-3.)
- A. Square wave _____ dB
 - B. Pulse wave _____ dB
 - C. Triangle wave _____ dB
 - D. Sawtooth wave _____ dB
 - E. Rectified sine wave _____ dB

CHAPTER 6 — Frequency and Power Distribution

INTRODUCTION

This chapter introduces you to waveform distortion. Distortion is a change in waveshape (waveform) that occurs when signals pass through imperfect electronic circuits. When circuits distort signals, they often create undesirable harmonic frequencies that impair the accurate reception of voice information. These circuits can also create harmonic signals at frequencies that may interfere with other telecommunication signals. The measure of distortion is one important factor in identifying the quality of communication signals.



OBJECTIVES

When you complete this chapter, you will be able to:

- Define the harmonic structure of common signal waveforms, and identify their frequency and power distribution.
- Specify the common forms of waveform distortion applied to signals that pass through electronic circuits.
- Specify the effects of circuit bandpass and frequency response upon signal waveforms and their harmonic structure.

THE SPECTRA OF COMPLEX WAVEFORMS

You have already learned that complex waveforms include harmonic frequencies within the wave shapes of fundamental frequency signals. You learned also that the specific wave shape of a signal depends upon the number, distribution, and power levels of its harmonics. The range of frequencies occupied by these harmonic signals is called the signal's "spectrum." You will now study the frequency distribution, the *spectra*, of harmonic signals. You will then study how these signals are used and controlled in audio and telephone systems.

Significant Power Levels

All non-sine waveforms have harmonics. Theoretically, *all* harmonic numbers (even numbered, and odd numbered) exist within the structure of complex waveforms; and there is an infinite number of these harmonics. We are usually concerned *only* with harmonics that have power levels of *significant amplitude*. So when listing the significant harmonics of any specific waveform, it is presumed that they are *at least* at a power level that can affect the circuits through which they pass.

Different electronic communications fields each have their own standards for which power levels are significant in specific circuits. These power levels are referenced and specified in dB units. Harmonic signals are usually referenced to the level of the fundamental frequency signal. Harmonic signals 60 dB below the power level of the fundamental are most often considered to be insignificant in audio and telephone practice. Harmonics 40 dB down will have no noticeable effect on an oscilloscope display of a signal's waveform, but they are usually considered to be significant. You will use 40 dB as your "threshold" signal level in this chapter.

The following equation, is used to compute power ratios from dB values:

$$\frac{P_2}{P_1} = \log_{10} \left(\frac{\text{dB}}{10} \right)$$

Using this power ratio equation, the power levels of harmonic signals that are 40 dB below the level of the fundamental calculate at a ratio of 1:10,000. This 40 dB ratio will allow you to identify which, and how many harmonics in any signal waveform are significant.

Using the “short-cut” technique you learned in the first unit, you should convert 40 dB to four decimal places (one decimal place for every 10 dB), to arrive at a 1:10,000 ratio.

10 = 1 and one 0 = 1:10

20 = 1 and two 00 = 1:100

30 = 1 and three 000 1:1000

Closely examine the harmonic structure of the signal shown on the oscilloscope and spectrum analyzer displays of Figure 6-1. By placing a line on the spectrum analyzer display at 40 dB below the level of the fundamental, you can: (1) identify the significant harmonic numbers, and (2) determine the range of frequencies, the spectrum, occupied by the signal.

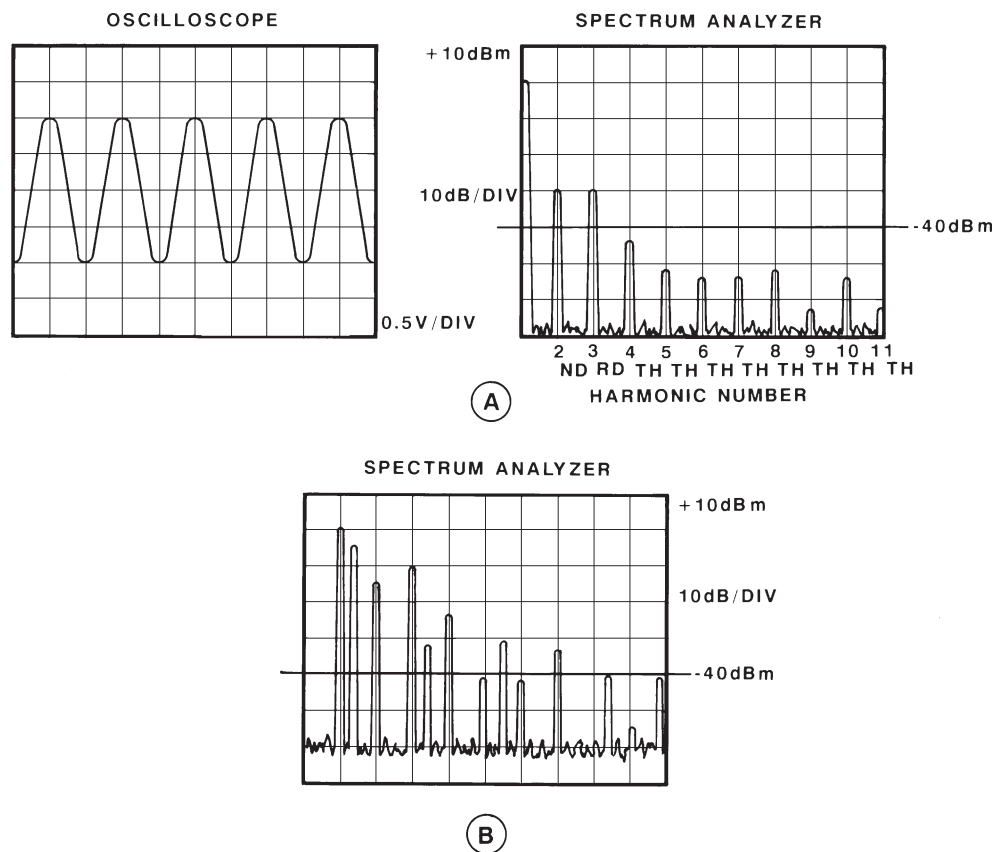


Figure 6-1

Oscilloscope and spectrum analyzer displays.

Figure 6-1A shows the displays of a sine waveform (not a perfect sine wave), with second and third harmonics at about 30 dB below the fundamental. All of the higher-order (higher-numbered) harmonics are at power levels of 45 dB or more, below the fundamental. Figure 6-1B shows the displays of a rectified sine waveform. This display shows the second, third, fourth, fifth, sixth, eighth, and tenth harmonics at power levels less than 40 dB below the fundamental. All other harmonics are below the 40 dB reference line.

Frequency Distribution

The frequency distribution of a signal is the range of frequencies occupied by the signal and its *significant* harmonics (harmonics less than 40 dB below the power of the fundamental). Frequency distribution is simply calculated as:

$$f_{\text{dist}} = \text{highest harmonic number} \times f_{\text{fund}}$$

For example, the frequency of the fundamental signal in Figure 6-1 is 400 Hz. For the waveform of Figure 6-1A, its frequency distribution extends through the third harmonic. (All harmonic signals above the third have power levels at more than 40 dB below the level of the fundamental and are therefore considered insignificant.) Its frequency distribution is:

$$\begin{aligned} f_{\text{dist}} &= \text{highest harmonic number} \times f_{\text{fund}} \\ &= 3 \times 400 \text{ Hz} = 1200 \text{ Hz} \end{aligned}$$

For the waveform of Figure 6-1B, its frequency distribution extends through the tenth harmonic. Its frequency distribution is:

$$\begin{aligned} f_{\text{dist}} &= \text{highest harmonic number} \times f_{\text{fund}} \\ &= 10 \times 400 \text{ Hz} = 4000 \text{ Hz} \end{aligned}$$

Signal frequency distribution is the range of frequencies occupied by a signal and its significant-level harmonics.

BANDWIDTH AND BANDPASS

From your review of Figure 6-1, you should conclude that waveforms with a greater number of significant harmonics occupy a wider range of frequencies. The circuits through which signals with complex waveforms pass, must be able to accommodate the range of distributed frequencies within the spectrum of that signal's waveform. The perfect sine wave will occupy only its frequency. The sine wave will thus require only limited frequency response within circuits that pass its waveforms. Most communication signals, and *especially* voice and audio signals, are complex, with significant harmonics. These signals require that circuits have extended frequency response in order to pass their waveforms.

For example, the extreme musical range of *fundamental* frequencies in the human voice extends from about 100 Hz for the lowest bass, to nearly 1000 Hz for the highest soprano. The frequency response of a voice channel of a telephone circuit extends over a range of about 3000 Hz. The frequency response for the sound information in radio broadcast signals will be at least 5000 Hz for AM stations, and 15,000 Hz for FM stations. The reason these communication services have extended frequency response, is to pass complex waveforms. The greater frequency response systems will thus provide the most accurate communication and reproduction of signal waveforms.

The *spectrum* or range of frequencies occupied by a signal waveform is called the signal's "bandwidth." The frequency response of the circuits through which these signals pass is called the circuit's "bandpass." Referring back to Figure 6-1A, the bandwidth (including harmonics to -40 dB) of the 400 Hz sine wave signal ranges from 400 Hz to 1200 Hz, a total of 800 Hz. A circuit with an 800 Hz bandpass will pass this signal accurately. For Figure 6-1B, the bandwidth extends from 400 Hz to 4000 Hz, a total of 3600 Hz. A circuit with 3600 Hz bandpass will be required to accurately pass this harmonically rich signal.

Bandwidth is the range of frequencies occupied by a signal's waveform. Bandpass is the range of frequencies that will pass through a circuit.

Look at Figure 6-2 and consider the example of a soprano, singing a note at 900 Hz, and with a waveform that includes harmonic signals through the tenth harmonic. The bandwidth of her voice waveform extends from 900 Hz through 9000 Hz, a range of 8100 Hz. The soprano's voice will not be accurately reproduced through telephone circuits or AM radio circuits, because of their limited bandpass. (All of the harmonics of the soprano's voice will not pass through the circuits that have a frequency response less than 900 to 9000 Hz.) You may understand the words she is singing, and you can accept the sound received through a telephone, or AM radio, as accurate communication. However, the most accurate communication will result when you hear the soprano's voice received through the wideband FM radio circuits.

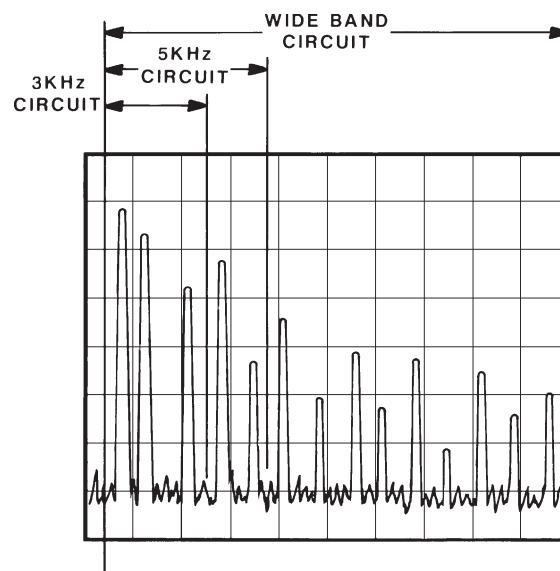


Figure 6-2

Limits of circuit bandpass on harmonic spectrum.

HARMONIC DISTORTION

“Distortion” is the word used to identify the extent to which signal waveforms may be changed as they pass through electrical or electronic circuits. A change in signal waveform is usually considered undesirable, as such changes will alter the character of communication signals. Harmonic distortion is the most common measure of circuit performance within communication systems. However, this type of distortion only indicates effects which add harmonics to signals, not those which remove harmonics from signals.

Harmonic distortion is the measure of power in the harmonics of a signal's waveform.

The measure of harmonic distortion is based upon the ideal perfect sine wave. The *total* power ($P_{\text{fund}} + P_{\text{har}}$) in a waveform's fundamental and harmonics is compared to the power in the harmonics alone (P_{har}). The ratio of these two power values is the distortion factor. Harmonic distortion is usually expressed as a percentage; sometimes abbreviated T.H.D. (for total harmonic distortion).

$$\text{T.H.D.} = \frac{P_{\text{har}}}{(P_{\text{fund}} + P_{\text{har}})} \times 100$$

Figure 6-2 repeats the two signals of our earlier examples of Figure 6-1, but identifies the power levels of all the signals on the spectrum analyzer display. For the sine wave of Figure 6-1A, the fundamental is at a power level of 1 watt; while the significant second, and third, harmonics are at a level of 1 milliwatt (30 dB below 1 watt). The power in each of the harmonics will add; so there are two milliwatts (.002 watts) in the harmonics of this waveform. The harmonic distortion for this waveform can be calculated as:

$$\text{T.H.D.} = \frac{P_{\text{har}}}{(P_{\text{fund}} + P_{\text{har}})} \times 100$$

$$\text{T.H.D.} = \frac{.002\text{W}}{(1\text{W} + .002\text{W})} \times 100 = 0.2\%$$

By contrast, the rectified waveform of Figure 6-1B will have a substantial amount of power in its harmonics; and a high harmonic distortion factor. With the same 1-watt power level of the fundamental, the power levels of the harmonics are as follows:

second harmonic $-10 \text{ dBW} = 0.1 \text{ W} = 100 \text{ mW}$

third harmonic $-20 \text{ dBW} = .01 \text{ W} = 10 \text{ mW}$

forth harmonic $-14 \text{ dBW} = .04 \text{ W} = 40 \text{ mW}$

fifth harmonic $-34 \text{ dBW} = .0004 \text{ W} = 0.4 \text{ mW}$

sixth harmonic $-24 \text{ dBW} = .004 \text{ W} = 4 \text{ mW}$

eighth harmonic $-30 \text{ dBW} = .001 \text{ W} = 1 \text{ mW}$

tenth harmonic $-34 \text{ dBW} = .0004 \text{ W} = 0.4 \text{ mW}$

The sum total of the power in the harmonics of this waveform is 0.1558 watts. The harmonic distortion for this waveform is calculated as:

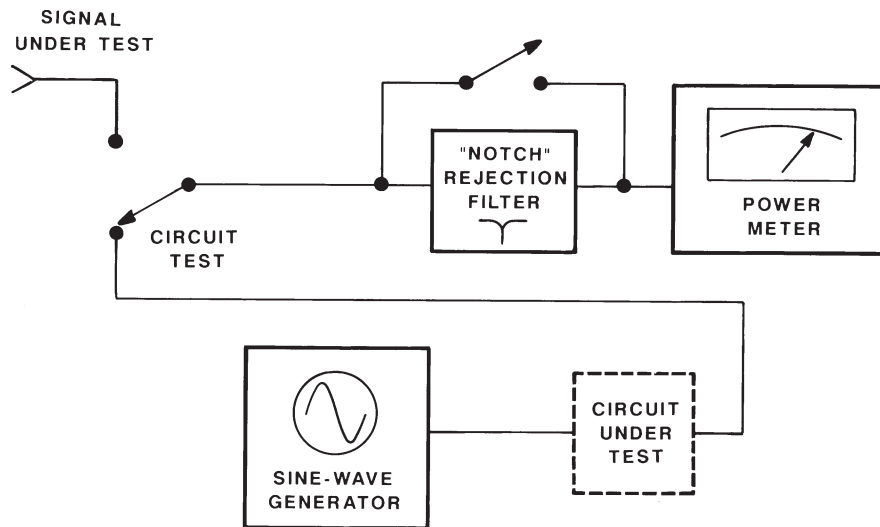
$$\text{T.H.D.} = \frac{P_{\text{har}}}{(P_{\text{fund}} + P_{\text{har}})} \times 100$$

$$\text{T.H.D.} = \frac{0.1558\text{W}}{(1\text{W} + 0.1558\text{W})} \times 100 = 13.48\%$$

For this calculation, you should observe that the second, third, and forth harmonics are the most significant, and account for 0.15 watts of the total 0.1558 watts (about 96%) of harmonic power.

Harmonic distortion can be precisely measured with the spectrum analyzer, but it is often measured with a lower-cost instrument called a “harmonic distortion analyzer.” This instrument is used both to test the harmonic content of signal waveforms, and to test the performance of audio, telephone, and communications circuits. The distortion analyzer is diagrammed in Figure 6-3.

The harmonic distortion analyzer is a power meter that includes a frequency-selective filter to remove the fundamental frequency from the waveform. The distortion is thus measured directly as the power ratio of the signals with and without the fundamental. The distortion analyzer also includes a sine-wave signal source to test the distortion affected as a signal passes through a circuit under test. In operation, the waveform power levels are measured at the circuit’s output, with and without the fundamental.

**Figure 6-3**

Harmonic distortion analyzer.

You can see that while the harmonic distortion factor is a measure of the harmonic content of a signal, it is also a measure of a circuit's performance. While waveforms do have harmonics, the concept of distortion is the addition of unwanted harmonics by the circuits through which signals pass. An ideal circuit will preserve the harmonic structure of any signal passing through it. Any change, or distortion, of a signal's waveform will change its harmonic structure; and thus alter the character of the signal. This change in waveform can adversely affect the accuracy of the information carried by the signal.

Two distinct factors that affect signal waveforms have been introduced in this section. One factor relates to the removal or suppression of harmonics, because of limited bandpass frequency of the circuits through which the signals pass. The result of this action is to change the distinctive harmonic structure of a signal's waveform. In practice, this sort of change in a signal's waveform is called "filtering", not distortion.

The second factor of change to a signal waveform relates to the addition of harmonics to a signal as it passes through a circuit. This effect will also change a signal's distinctive harmonic structure. This kind of change to a signal's waveform is called "harmonic distortion."

SUMMARY

The specific wave shape of a signal depends upon the number, distribution, and power levels of its harmonics. Although there is an infinite number of harmonics in a complex waveform, only those that have power levels of significant amplitude are listed as “significant harmonics.” The bandwidth of a signal is the range of frequencies occupied by a signal and its significant harmonics.

Harmonic distortion is the measure of power in the harmonics of a signal’s waveform. While waveforms do contain harmonics, harmonic distortion is the unwanted harmonics added to signals. Harmonic distortion is also a measure of a circuit’s performance.

SELF-TEST REVIEW

1. The fundamental frequency of a signal is at a power level of 1 watt. At which power level are the signal’s harmonics “insignificant?”
 - A. 10 mW
 - B. 1 mW
 - C. 0.1 mW
 - D. .01 mW
2. A rectified sinewave signal (refer back to Figure 5-3 on Page 5-9) will include its first two harmonics within its waveform. What is the highest harmonic number in the waveform?
 - A. second.
 - B. third.
 - C. fourth.
 - D. fifth.
3. A square wave signal (refer back to Figure 5-3 on Page 5-9) will include its first two harmonics within its waveform. What is the highest harmonic number in the waveform?
 - A. second.
 - B. third.
 - C. fourth.
 - D. fifth.

4. What is the frequency distribution of a 400 Hz signal if its highest significant harmonic number is 5? _____
5. The significant harmonics of a 400 Hz signal extend through its seventh harmonic. What is the bandwidth of the signal?
 - A. 2000 Hz
 - B. 2400 Hz
 - C. 2800 Hz
 - D. 3200 Hz
6. A 400 Hz signal with many significant harmonics is applied to a circuit that has a bandpass of 1700 Hz, beginning at 350 Hz and extending through 2050 Hz. What is the highest harmonic number that will pass through the circuit?
 - A. third.
 - B. fourth.
 - C. fifth.
 - D. sixth.
7. The total power in the fundamental frequency of a complex waveform is 1 watt. The total power in its harmonics is 0.1 watt. What is its T.H.D.?
 - A. 8%
 - B. 9%
 - C. 10%
 - D. 11%
8. A signal waveform is “distorted” when passing through a circuit, in a way that reduces its harmonics. What is the correct name for this type process? _____
9. Refer to Figure 6-1B, and note the number of harmonics less than 30 dB below the power level of the fundamental. _____
10. Refer to Figure 6-1B again, and determine the bandwidth of the signal that includes all harmonics less than 30 dB below the power level of the fundamental. The frequency of the fundamental is 400 Hz. Record your answer in Hertz. _____ Hz
11. A spectrum analyzer display shows the second harmonic of a signal to be at 26 dB below the 1-watt level of the fundamental. What is the power level of the second harmonic? _____ mW

12. Refer to Figure 6-1B, and list the power levels of each of the harmonics less than 30 dB below the power level of the fundamental. Assume that the power level of the fundamental is 2 watts; then translate each of the harmonic power levels to milliwatts, and calculate the T.H.D.

1. _____.

2. _____.

3. _____.

4. _____.

T.H.D. _____ %

13. A complex waveform of a signal has 800 mW in its fundamental. The power level of the signal's harmonics are as follows:

200 mW in its 2nd. harmonic

100 mW in its 3rd. harmonic

50 mW in its 4th. harmonic

20 mW in its 5th. harmonic

Calculate the T.H.D. of the signal waveform. _____ %

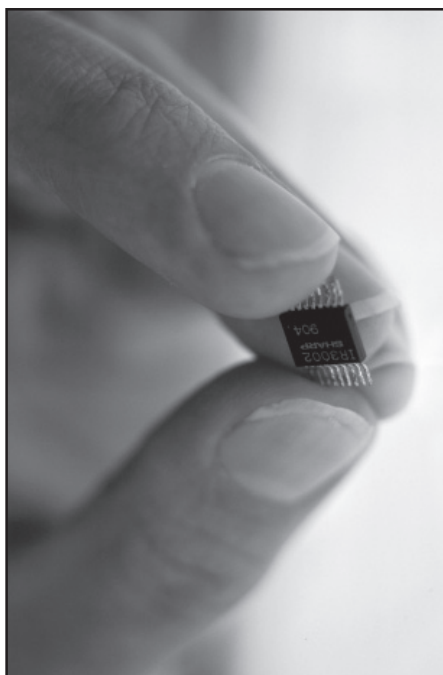
14. The signal described in question 13 is passed through a filter that suppresses all of the harmonic signals but the second. What is the T.H.D. for the signal with suppressed harmonics? _____ %

CHAPTER 7 — Filters

INTRODUCTION

Almost all communications circuits are designed to provide the best frequency response for the specific information they process. The deliberate control of a circuit's frequency response is most often accomplished with the use of filters. Filters are used to limit the frequency response of circuits. They do this by passing specific frequencies within the spectrum, and preventing all others from passing through circuits.

Filters are made using frequency selective components. Most simple filters are comprised of *electrical* networks that combine inductance and/or capacitance with circuit resistance. There are many filter circuits that use *electronic* components and IC devices in their frequency-selective networks. There are also a number of *acoustic*, piezoelectric and other resonating elements, that are used in filter circuits. These unique acoustic components are made of natural quartz, synthetic crystal, ceramic, and other special materials.



In this chapter, you will study the electrical properties of filters. Later, you will learn how filters are used to limit the frequency response of circuits; and how filters are used to select and reject specific signal frequencies.

OBJECTIVES

When you complete this chapter, you will be able to:

- Determine the effects of selective filtering on the suppression and/or selection of specific harmonic signals.
- Relate the concepts of harmonic distortion and frequency generation to telephone, radios, and telecommunication systems.

FILTER CHARACTERISTICS

The words “bandwidth,” “bandpass,” “passband,” “stopband,” “octave,” “skirt,” “slope,” “selectivity,” “rejection,” and the abbreviation “Q,” all relate to the frequency response of filters. You will likely hear these words used both correctly and incorrectly in practice.

The terms *passband*, *bandpass*, and *bandwidth* are often very confused and misused. The word “*passband*” refers to a range of *signal* frequencies. The word “*bandpass*” refers to the frequency range of the *filter* (or other circuit). The word “*bandwidth*” usually refers to the *spectral distribution* of a signal—the range of frequencies occupied by a signal and its harmonics. However, bandwidth and bandpass are very often used interchangeably as accepted terms in describing the frequency response of filters. Our concern here is that you understand the difference between them.

For example, let’s consider a filter that has a signal *passband* from 2 kHz to 7 kHz. Its *bandpass* will be 5 kHz (7 kHz – 2 kHz). Now let’s apply a 2 kHz complex waveform signal with many harmonics to the filter. This signal has a waveform with significant harmonics that extend through 14 kHz. The signal’s bandwidth is 12 kHz (14 kHz – 2 kHz) at the filter’s input. However, only the fundamental (2 kHz), second harmonic (4 kHz), and third harmonic (6 kHz), of the signal’s *bandwidth* will pass through the filter’s limited *bandpass* of 5 kHz. The signal’s *bandwidth* will have been reduced to 4 kHz (6 kHz – 2 kHz) at the filter’s output.

“Passband” refers to a range of *signal* frequencies.
“Bandpass” refers to the frequency range of *circuits*.
“Bandwidth” properly refers to the *spectral distribution* of a signal, but is often used to specify circuit bandpass.

Filters are specified by the frequencies where their properties change. These frequencies are usually indicated where the filter’s output signal level is reduced to 3 dB below the level of its passband frequencies. The bandpass of filters is also indicated by the *range* of frequencies within their passband; and this range is often expressed in octaves. (An “octave” is a 2:1 frequency ratio.) If the passband is less than an octave, the range is most often expressed in percentage of an octave. (A 10% bandpass is 10% of an octave, or 0.1 octaves.)

If a filter passes frequencies from 1 kHz to 8 kHz, for example, it has a bandpass of three octaves: 1 – 2 kHz is one octave, 2 – 4 kHz is the second octave, and 4 – 8 kHz is the third octave. The bandpass (BP) of the filter, in octaves, can be calculated with the following formula:

$$BP_{(\text{octaves})} = \frac{\log \left(\frac{f_{\text{hi}}}{f_{\text{low}}} \right)}{0.3}$$

A wideband filter with a passband from 200 Hz to 8 kHz, for example, has a bandpass, expressed in octaves, of:

$$BP_{(\text{octaves})} = \frac{\log \left(\frac{8000 \text{ Hz}}{200 \text{ Hz}} \right)}{0.3} = 5.34 \text{ octaves}$$

A narrow-band filter with a passband from 200 Hz to 300 Hz, as another example, has a percentage bandpass of:

$$BP \% = \frac{\log \left(\frac{300 \text{ Hz}}{200 \text{ Hz}} \right)}{0.3} \times 100 = 58.7\% \text{ (of an octave)}$$

Note that we have now presented two different equations specific to bandpass. The first is used to calculate the bandpass in octaves. The second is used to calculate the bandpass in percent of an octave. Be sure you understand the desired result when you are asked to calculate bandpass.

The amount by which signal power is reduced at any frequency beyond (or outside of) the filter's bandpass is called "rejection." Filter rejection is a power loss, which is specified and measured in dB. Refer to Figure 7-1, and note the terms applied to the graph of filter frequency response. Note that the frequency range where rejection occurs is identified as the "skirt" (or "skirts") of the filter's response. Since the signals are rejected, rather than passed within this range of frequencies, the range is also called a filter's "stopband."

The change in rejection with frequency on the skirt of the filter's response results in a slope on the graph. Filters with steeper slope curves will have a greater change in output with a given change in frequency. Such filters will be more responsive, more "selective", to changes in frequency. Therefore, a filter rated to have high selectivity, has steep slopes in its stopband frequency response curves.

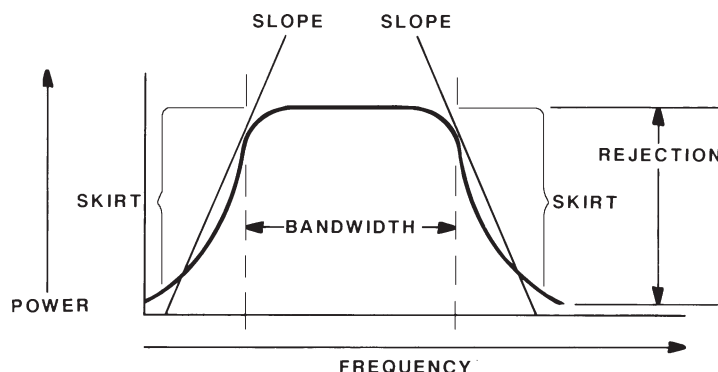


Figure 7-1

Filter terminology.

The filter reduces signal power on the *skirts* of its response. The amount by which the power is reduced at any frequency on the filter's skirt is its "rejection." The change in power vs. frequency on the skirt is the filter's "slope."

FILTER TYPES

There are four general categories of filters: low-pass, high-pass, bandpass, and bandstop. As their names imply, these filters *pass* a range of frequencies, while *stopping* another. Figure 7-2 shows examples of the four filter types, and their general frequency response. The low-pass filter of Figure 7-2A is specified at 8 kHz. It will pass all frequencies below 8 kHz, and its power output at 8 kHz will be 3 dB below the level of its passband signals. The signal power passing through the filter *above* 8 kHz will be drastically reduced.

The high-pass filter of Figure 7-2B is also specified at 8 kHz. However, its passband is above 8 kHz, and extends to some frequency limit beyond the range of the higher-frequency signals it is intended to pass. When low-pass and high-pass filters are combined, they can provide a wide passband limited at both low and high frequencies. The example of Figure 7-2C shows a 200 Hz high-pass filter, combined with an 8 kHz low-pass filter to create a 5.34 octave bandpass (a passband from 200 Hz to 8 kHz).

When filter response is limited in both high-pass and low-pass frequency limits, the filter is called a “bandpass” filter. A filter limiting a specific frequency or range of frequencies is called a “band stop” filter. This is shown in Figure 7-2D.

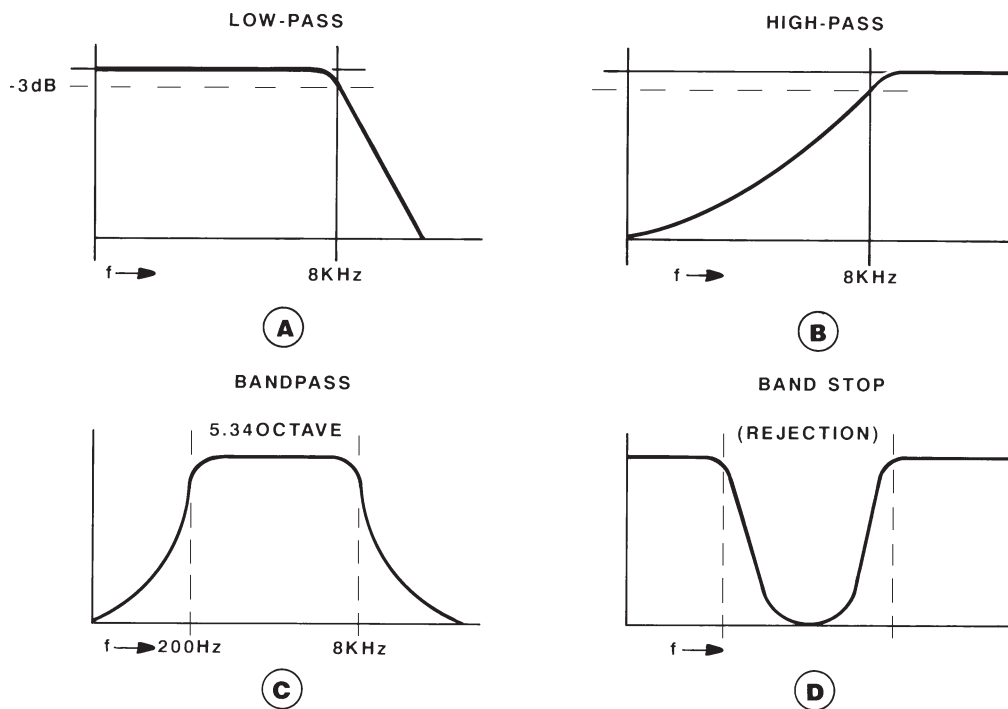


Figure 7-2

Frequency response of four filter types.

Narrow-band filters of about 25%, or less, of an octave can be obtained using simple LC resonant networks. Figure 7-3A shows a typical LC bandpass filter network; while Figure 7-3B shows a typical LC band stop filter network. These networks are specified for bandpass properties (bandwidths) that are proportional to the ratio of their respective resistance and reactance values. This ratio is called “Q”, for quality factor. Q is a measure of selectivity, the ratio of a change in power vs. a change in frequency on the slopes of a filter’s frequency response curve. The relationships between the frequency-limit specifications for these narrow-band filters can be calculated with the following equations:

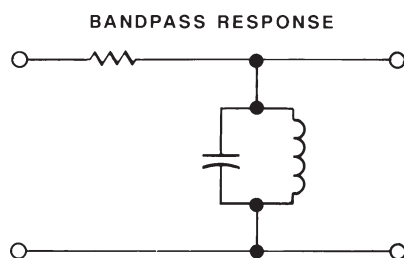
$$Q = \frac{X}{R}$$

$$BP = \frac{f_0}{Q} \quad \text{or} \quad BP = f_{hi} - f_{low}$$

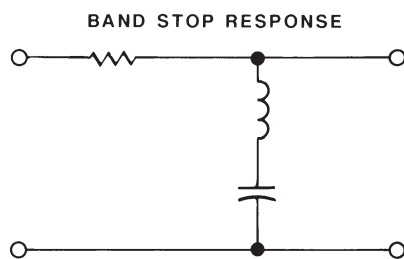
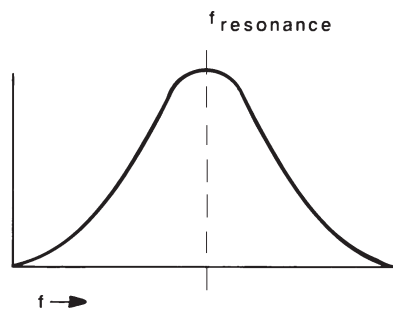
$$f_{hi} = f_0 \left(1 + \frac{1}{2Q}\right)$$

$$f_{low} = f_0 \left(1 - \frac{1}{2Q}\right)$$

$$BP\% = \frac{\log\left(\frac{f_{hi}}{f_{low}}\right)}{.03} \times 100$$



A



B

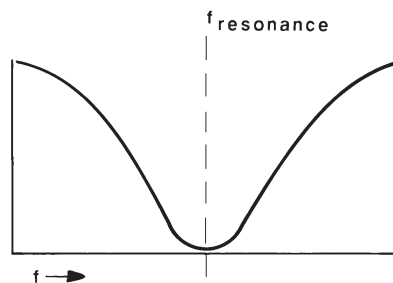


Figure 7-3
Basic LC filters.

For these equations, f_0 is the center (resonant) frequency of the network, where $X_C = X_L$, X is the reactance (X_C or X_L) and R is the network resistance (including the resistance distributed in the windings of the inductor).

As an example, let's calculate the properties of a simple resonant network to be used as either a band-pass or band-stop filter. If we assume a reactance of 200 ohms and a resistance of 10 ohms, at a frequency of 1 kHz, we can compute the filter's properties as follows:

$$Q = \frac{X}{R} = \frac{200}{10} = 20$$

$$BP = \frac{f_0}{Q} = \frac{1 \text{ kHz}}{20} = 50 \text{ Hz}$$

$$f_{hi} = f_0 \left(1 + \frac{1}{2Q} \right) = 1 \text{ kHz} \left(1 + \frac{1}{2 \times 20} \right) = 1025 \text{ Hz}$$

$$f_{low} = f_0 \left(1 - \frac{1}{2Q} \right) = 1 \text{ kHz} \left(1 - \frac{1}{2 \times 20} \right) = 975 \text{ Hz}$$

$$BP\% = \frac{\log \left(\frac{f_{hi}}{f_{low}} \right)}{0.3} \times 100$$

$$BP\% = \frac{\log \left(\frac{1025}{975} \right)}{0.3} \times 100$$

$$BP\% = 7.24\% \text{ (of an octave)}$$

EQUALIZATION NETWORKS

When it is applied to telecommunication practice, the word equalize means to compensate. There are a number of factors that affect the frequency response of electronic components and circuits. When signals pass along circuit paths, for example, capacitance causes increased losses at higher frequencies. If such losses are anticipated, equalization networks are connected to these circuits to *compensate* or *equalize* them. The result is a circuit with uniform frequency response. The equalization network corrects for, or otherwise adjusts, the frequency response of the circuit.

Equalizing networks are filters used to compensate for power variations in the frequency response of communication circuits.

There are many undesirable frequency response characteristics inherent within the circuits and cables that process communication signals. Filters can be used to compensate for these problems. When filters are used in this way, they are called "*equalization networks*" or "*equalization filters*." Figure 7-4 graphs the undesirable loss in telecommunication transmission circuits that use coaxial cables. This loss is predictable, and results from using specific cable lengths in standardized systems.

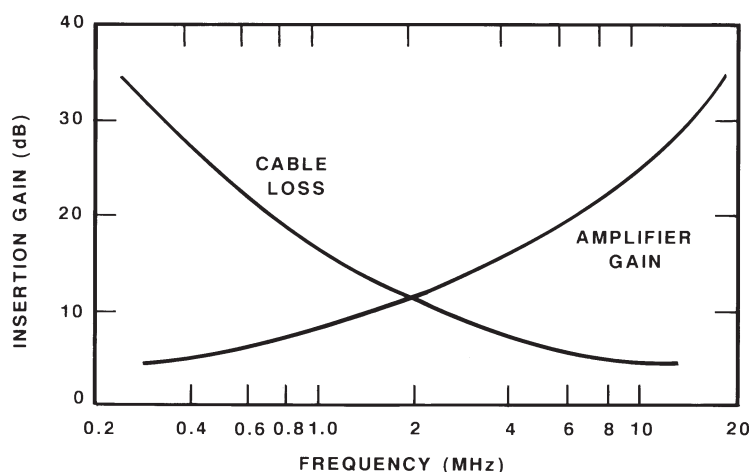


Figure 7-4

Cable loss vs. equalizing amplifier gain.

The figure also shows the response of a likewise standardized equalizing network, which is used to compensate the circuit. The equalizing network includes a high-pass filter and an amplifier. The signals operate at the skirt frequencies of the filter, where the filter's slope is opposite to that of the cable's frequency response. The filter's slope increases with frequency. The gain of the amplifier is set to compensate for the power loss of the signal. Equalizing amplifiers of this type are placed at convenient locations along the cable circuit. Their gain and equalization slope are often adjustable to compensate for cable runs of different lengths. The slope of these equalizing amplifiers is often called amplifier "tilt."

The frequency response of a telephone voice channel is standardized for efficient communication of speech and low-speed data signals. The standards are applied to both the telephone set, and to the circuits to which the telephone interconnects. In this instance, equalizing filters are used to *purposefully* limit and shape the frequency response of the equipment and circuits. For telephone equipment, this type of equalization is called "weighting."

Figure 7-5 shows the effects of equalizing filters upon the frequency response of a telephone circuit. The response curve is standardized so that all telephone circuits operate with similar performance. The particular frequency response curve shown in Figure 7-5, is called "C message weighting." This frequency response is obtained by using filter networks with both low-pass and high-pass characteristics, and with a peak in response at 1 kHz.

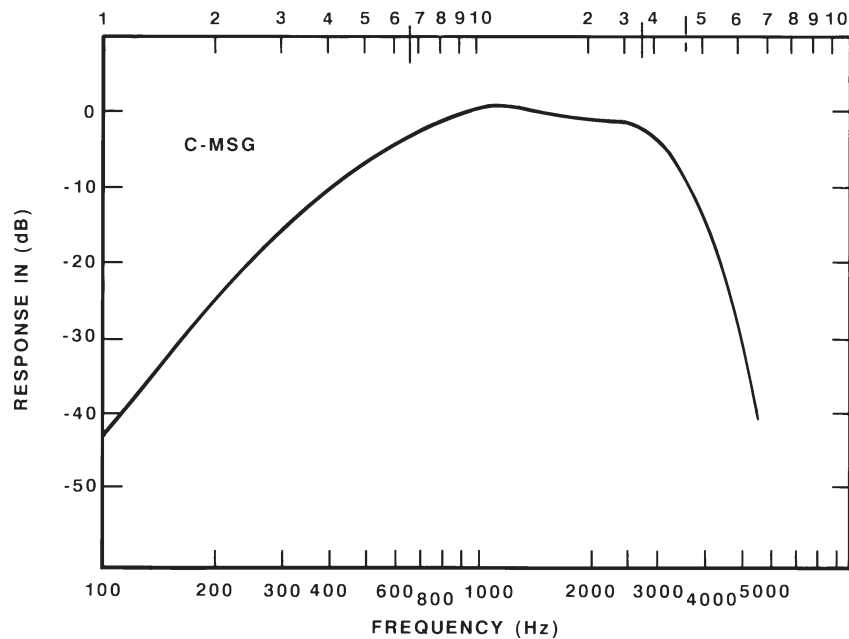


Figure 7-5

C-message weighted telephone frequency response.

You can learn much about the equalizing filter of Figure 7-5 by examining the graph. You should note that the slope (selectivity) of the filter's lower-frequency skirt is not as steep as the slope of its high-frequency skirt. By estimating the logarithmically spaced frequencies on the X-axis of the graph, you should observe that the filter's rejection is 10 dB at about 400 Hz and 4600 Hz. Your estimate of the filter's passband, at its -3 dB response levels, should extend from about 660 Hz to 3500 Hz, a bandpass of 2840 Hz. The response of the filter, from about 200 Hz to 400 Hz (an octave), varies from -26dB to -10 dB. Its slope, and selectivity, is 16 dB per octave.

At its high-frequency skirt, the response of the filter varies from -10 dB at about 4600 Hz, to -40 dB at 6600 Hz. This frequency range is less than an octave. You can determine the slope per octave with the same formula used to calculate bandpass:

$$BP\% = \frac{\log\left(\frac{f_{hi}}{f_{low}}\right)}{0.3} \times 100$$

$$BP = \frac{\log\left(\frac{6600}{4600}\right)}{0.3}$$

$$BP = 0.52 \text{ octaves}$$

The slope (selectivity) of the high-frequency skirt of the filter ranges from -10 to -40 dB, for a variation of 30 dB in 0.52 octaves. The slope in dB-per-octave can now be calculated by proportionality, as:

$$\text{Slope} = \text{dB variation} \times \frac{1}{\text{octave}}$$

$$\text{Slope} = \frac{30 \text{ dB}}{0.52 \text{ octaves}}$$

$$\text{Slope} = 57.7 \text{ dB/octave}$$

SUMMARY

Communication circuits are designed to provide the best frequency response for the specific information they process. Filters are used to control the circuit's frequency response. There are four general categories of filters: low-pass, high-pass, bandpass, and bandstop.

The terms passband, bandpass, and bandwidth are often misused. Passband refers to a range of signal frequencies. Bandpass refers to the frequency range of the filter. Bandwidth refers to the range of frequencies occupied by the signal and its significant harmonics.

SELF-TEST REVIEW

1. A filter has a passband from 500 Hz to 2000 Hz. What is its bandpass?
 - A. 500 Hz
 - B. 1000 Hz
 - C. 1500 Hz
 - D. 2000 Hz
2. A filter has a bandpass of two octaves. Its lower frequency limit is 500 Hz. What is its higher frequency limit?
 - A. 500 Hz
 - B. 1000 Hz
 - C. 1500 Hz
 - D. 2000 Hz
3. The complex waveform of a 500 Hz signal includes its second and third harmonics. What is the bandwidth of the signal?
 - A. 500 Hz
 - B. 1000 Hz
 - C. 1500 Hz
 - D. 2000 Hz

4. The power at one frequency point on the skirt of a filter's response is 100 mW at a frequency of 500 Hz. The power at a second point on the skirt is 10 mW at a frequency of 1000 Hz. What is the filter's slope (selectivity) in dB per octave?
 - A. 3 dB/octave
 - B. 6 dB/octave
 - C. 10 dB/octave
 - D. 20 dB/octave
5. Which type of filter will pass frequencies from DC to 5 kHz, but reject frequencies above 5 kHz?
 - A. bandpass
 - B. band-stop
 - C. low-pass
 - D. high-pass
6. Which type of filter will pass frequencies below 3 kHz and above 3.4 kHz, and reject the frequencies between these two limits?
 - A. bandpass
 - B. band-stop
 - C. low-pass
 - D. high-pass
7. Which type of filter would you use to suppress harmonics from a signal's waveform?
 - A. bandpass
 - B. band-stop
 - C. low-pass
 - D. high-pass
8. Which type of filtering will result if a parallel LC (inductance-capacitance) resonant network is placed across a circuit?
 - A. bandpass
 - B. band-stop
 - C. low-pass
 - D. high-pass

9. If you lower the value of the resistance in an LC resonant network, what will be the result?
- A. Reduced Q and higher selectivity.
 - B. Increased Q and higher selectivity.
 - C. Reduced Q and lower selectivity.
 - D. Increased Q and lower selectivity.
10. What will cause larger bandwidth in an LC resonant network?
- A. Reduced Q and higher selectivity.
 - B. Increased Q and higher selectivity.
 - C. Reduced Q and lower selectivity.
 - D. Increased Q and lower selectivity.
11. An equalizing amplifier is designed to compensate for transmission cable loss, due to long cable runs. The amplifier's frequency response is called:
- A. "Tilt"
 - B. "Slope"
 - C. "Weight"
 - D. "Load"
12. Calculate the bandpass of a filter, in octaves, if its frequency limits are at 400 Hz and 3400 Hz. _____ octaves
13. Calculate the percent of an octave bandpass of a narrow-band filter that operates between 56 kHz and 80 kHz. _____ %(octave)
14. A series resonant LC network has 1000 ohms reactance, and 20 ohms resistance. It is resonant at 3000 Hz. Calculate the Q and bandpass of the filter, and its passband range of frequencies.
- _____
- _____
15. Calculate the slope (or selectivity) of a filter, in dB per octave, if the rejection is 10 dB at 500 Hz, and 20 dB at 800 Hz. _____ dB/octave

CHAPTER 8 — Utilizing Waveforms and Harmonics

INTRODUCTION

In Chapters 6 and 7 the general term “distortion” was introduced. It was described as a change in the structure of a waveform that occurs as a signal passes through a circuit. Later, we made a distinction between harmonic distortion and filtering; and noted that “harmonic distortion” is the name given to the process of *adding* signals and harmonics to a waveform; while “filtering” is the name given to the process of *removing* signals and harmonics from a waveform.

While the effects of distortion are usually considered undesirable, waveform distortion is sometimes used to *create* harmonic frequencies.

To keep our intentions clear, this deliberate process is called “harmonic generation” or “frequency multiplication”, not signal distortion. Your study of this chapter will prepare you to understand these desirable forms of signal distortion.

You will also learn how filters are used in electronic communication systems.



OBJECTIVES

When you complete this chapter, you will be able to:

- Relate the results of waveform distortion to the generation of harmonic frequency signals.
- Apply concepts of harmonic generation and frequency response to the analysis of electronic comb and marker signal sources.
- Relate the concepts of harmonic distortion and frequency generation to telephone, audio, and telecommunication systems.

HARMONIC SUPPRESSION

Complex waveforms, with significant numbers of harmonics, require wideband circuits. Wideband circuits have extended bandpass properties that allow the higher-numbered harmonics to be maintained as signals pass through these circuits. Using wideband circuits poses a number of trade-offs for audio and telephone practice, however. You should understand these trade-offs and how they are used to advantage and disadvantage with communication signals.

The primary advantage of wideband circuits is their ability to *accurately* pass communication signals, by retaining all of the significant level harmonics in a signal's waveform. The disadvantages of wideband circuits are many, and it is for this reason that filters are used to limit circuit frequency response. The loss of significant harmonics is justified, because wideband circuits have limitations. These include:

1. **Limiting channel capacity.** It is possible to provide up *to five* 3 kHz telephone voice channels within the frequency bandpass of only *one* 15 kHz wideband music channel.
2. **Higher noise power.** The power level of noise within circuits is directly proportional to their frequency response, so the noise power will be five times higher with 15 kHz bandpass than it is with 3 kHz bandpass.
3. **Power inefficiency.** Signal power is distributed among the harmonics within a waveform. Signals with harmonics consume more power when they are processed through wideband circuits.
4. **Phase distortion.** The higher frequency harmonics take longer to reach a distant point in transmission circuits, causing phase distortion at the receiving end of the circuits.

Wideband circuits accurately pass signals and retain harmonics, but they have disadvantages that are avoided by narrow-band circuits.

In general, voice communication can be effective using narrow-band circuits that suppress higher-frequency harmonics. Figure 8-1 shows the power vs. frequency response of typical human speech, including harmonics. This figure compares the limited bandpass of a 3 kHz voice channel, and a 5 kHz audio circuit, with a wideband audio signal, such as broadcast from an FM radio station. Figure 8-1 shows most of the power in a voice signal within the range of the fundamental and lower numbered harmonics.

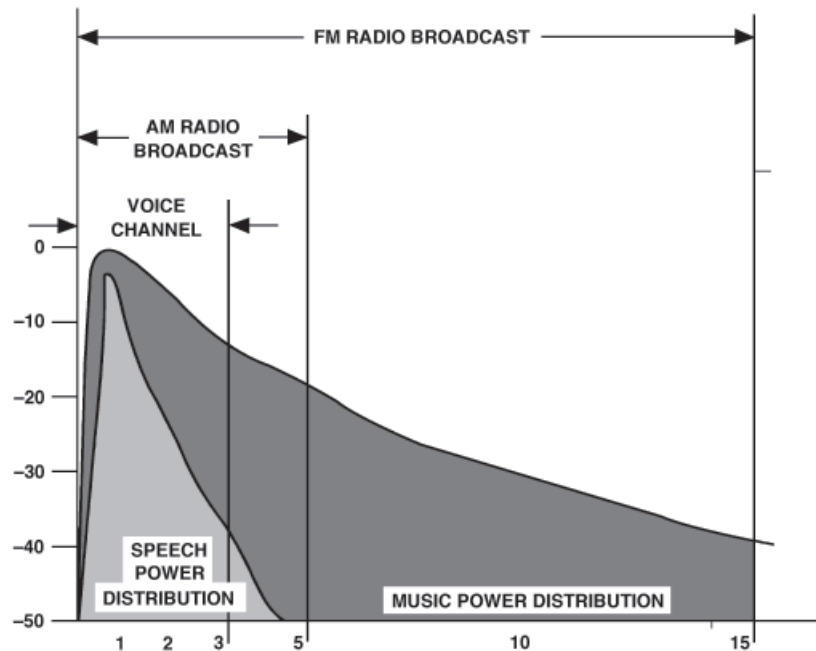


Figure 8-1

Power vs. frequency of voice and music signals.

Telephone circuits with about 3 kHz bandpass, are intended to communicate voice and low-speed data signals. With this limited frequency response, they suppress the higher-frequency harmonics from a signal's waveform. While this trade-off will reduce a circuit's ability to *precisely* reproduce a signal, reliable communication is still achieved. When the standards were established for telephone practice, the benefits of *narrow-band* operation were elected in order to avoid many of the problems associated with wideband circuits.

Filtering

Narrow-band circuits, such as those used for telephone voice channels, use filters to set their frequency response and specific passband limits. In this application, the filters alter the waveshape of signals with many harmon-

ics, and reduce them to simpler waveforms with fewer harmonics. Filter circuits with a very narrow frequency response will reduce virtually any complex waveform to a single fundamental-frequency sine wave.

You have likely heard electronic musical instruments, such as synthesizers and keyboards. Filtering for harmonic suppression is one method of obtaining different waveforms and sounds from these instruments. The same method is used in some large electronic church and concert organs. The fundamental frequencies of these musical instruments begin as harmonically rich complex waveforms. These waveforms initially simulate instruments with harsher sounds, such as bassoons and trumpets. When the waveforms pass through filters, their harmonics are suppressed, and they look and sound more like sine waves. Sounds from sine waveforms are mellow. Instruments with sine-like waveforms include flutes and acoustic guitars.

Using filters to suppress harmonics allows us to obtain pure-tone sine waves from complex signal waveforms. For example, the signals generated using electronic IC devices are seldom perfect sine waves. In some circuits, it is often more practical to use digital ICs to generate stable square wave signals. If sine waveforms are required, the square-wave signals are passed through high-Q (narrow-band) filters which suppress all of their harmonic frequencies.

Filters are used to reduce the harmonic content of signal waveforms. Filters can reduce very complex waveforms to single-frequency sine waves.

The technique of harmonic suppression by filtering is applied to almost all radio transmitters. The final amplifier in these systems is usually set for maximum power efficiency. This amplifier conducts during only a portion of the signal's cycle. The result is a rectified signal waveform with many harmonics.

If a rectified (complex) waveform is applied directly to an antenna, the transmitted signal will contain many harmonic frequencies. These harmonics may interfere with other radio and telecommunication services, and render them useless. With this potential for disastrous consequences, all radio transmitters must, by law, suppress all harmonics from their outputs. They must transmit sine waveforms at only the frequency for which they are licensed. This required harmonic suppression is accomplished in radio transmitters with extensive filtering.

You may have encountered the results of poor harmonic suppression from CB (U.S. Citizen's Band) radio transmitters. The operating frequency of CB transmitters is from 26.965 through 27.405 MHz. The second harmonic of this frequency range extends from 53.93 through 54.81 MHz, which is within the operating frequency of U.S. television Channel 2. The third harmonic extends from 80.81 to 82.215 MHz, within the operating frequency of U.S. television Channel 5 (and some overlap into Channel 6). A poorly filtered CB transmitter in your neighborhood will produce harmonics that can cause visible interference problems in television reception.

HARMONIC GENERATION

Harmonic generation is the technique of creating harmonics adding harmonics to an existing waveform. It is the opposite effect of the suppression of harmonics. There are many signal processing circuit functions used in communications systems that rely on the presence of harmonic signals. These circuits are different than those we described earlier, where suppressed harmonics were required.

Generation of harmonic signals requires complex waveforms. These waveforms can originate as triangle, square, or pulse signals, or they can be created by distorting sine wave signals. The electronic circuits that distort signals applied to their inputs are called "non linear" circuits. A linear circuit, by contrast, will pass a signal to its output with exactly the same waveform as it has at the circuit's input. A perfect sine wave will pass through a perfectly linear circuit, and have no harmonics at its output.

Linear circuits transfer signals without changing their waveforms. *Non-linear* circuits distort the waveforms of signals passing through them. The distortion of waveforms creates harmonics.

Specially designed non-linear electronic circuits allow us to obtain complex waveforms with desired harmonic properties. Earlier, we discussed the example of how some electronic musical instruments generate complex waveforms, and then filter them to suppress their harmonics. There are also many electronic musical instruments that create their complex waveforms in reverse. They initially generate mellow sounding sine waveforms. These sine waves are then passed through non-linear circuits to create the harsher-sounding complex waveforms of additional musical instruments.

The circuits that generate and process complex waveforms must have sufficient frequency response to pass all of that waveform's significant harmonics. Wideband circuits are thus required for all complex waveforms. The exact passband required for these signals depends upon the application (use) of the signals. The limits of frequency response are simply calculated from the highest fundamental frequency and harmonic number for any specific waveform.

Table 8-1 shows the power levels of significant harmonics for some popular waveforms. Table 8-1 can be compared with the graphs of Figure 5-3 on page 5-9. From your study of Table 8-1 (and Figure 5-3), you should note that a square-wave signal has significant harmonic levels (less than 40 dB below the fundamental) through the ninth harmonic. With a fundamental frequency range from 100 Hz to 1 kHz, for example, the passband of a circuit that retains this square waveform (and its harmonics) must extend from 100 Hz through 9 kHz ($9 \times 1 \text{ kHz}$).

Table 8-1

WAVEFORM	HARMONIC NUMBER								
	2	3	4	5	6	7	8	9	10
SQUARE	---	-9	---	-14	---	-16	---	-18	---
PULSE	-1	-2	-1	-4	-6	-9	-13	-20	-30
TRIANGLE	---	-18	---	-28	---	-30	---	-36	
SAWTOOTH	-6	-10	-13	-14	-15	-16	-17	-18	-19
RECTIFIED SINE	-3	---	-18	---	-22	---	-26	---	-28

In practice, the specific waveforms likely to be generated by a non-linear circuit determines its frequency response. Generally, frequency response through the ninth harmonic retains the waveforms of almost all communication signals. The decision to extend the frequency response of FM broadcast radio through 15 kHz, thus assures that the ninth harmonic of a complex waveform with a fundamental frequency of 1.6 kHz is retained within the waveform ($9 \times 1.6 \text{ kHz} = 14.4 \text{ kHz}$). Likewise, virtually any music waveform generated with a fundamental frequency up through 1.6 kHz will closely maintain its waveshape as is processed through 15 kHz bandpass circuits.

Harmonic Generators

There are circuits used in communication systems that generate and retain many signal harmonics. The frequency response of these circuits extends over an extremely wide range of frequencies. One of these circuits is the harmonic generator. Depending upon its application, the circuit may be called a “marker generator” or “comb generator.” This circuit produces many signals, incrementally spaced at its fundamental frequency. Once generated, the harmonic signals pass through an equalizing amplifier to bring their power to the same approximate level.

Harmonic generators are used to create signals equally spaced in frequency and at equal power levels across a range of frequencies.

A comb generator’s output looks like an inverted comb on a spectrum analyzer display, with each tooth of the comb representing one of the incrementally-spaced harmonic signals. Figure 8-2A shows a frequency-domain graph of the comb generator’s output signals.

There are many popular applications for harmonic generator circuits of this type. One is for the creation of frequency reference signals for electronic test equipment. With a very precise and stable fundamental frequency at 5 kHz, for example, the circuit provides exact reference signals for test purposes at 5 kHz increments. This type of signal is used for the CRT displays of network analyzers to graphically indicate frequency. Figure 8-2B shows the typical CRT display of these harmonic signals combined with the response curve of a circuit under test. When used for test purposes of this type, the harmonic signals are called “markers,” as they mark frequencies.

Another application for the harmonic generator is the creation of signals for multiplexing of telephone voice channels. Multiplexing provides for the simultaneous transmission and reception of many voice signals within a single circuit. This process is accomplished by generating many harmonic signals that will each carry voice information at a different frequency. The name given to this type of multiplexing is “frequency division multiplexing,” abbreviated FDM.

Frequency-Division Multiplexing places many voice signals on a single transmission circuit by spacing them at different carrier frequencies.

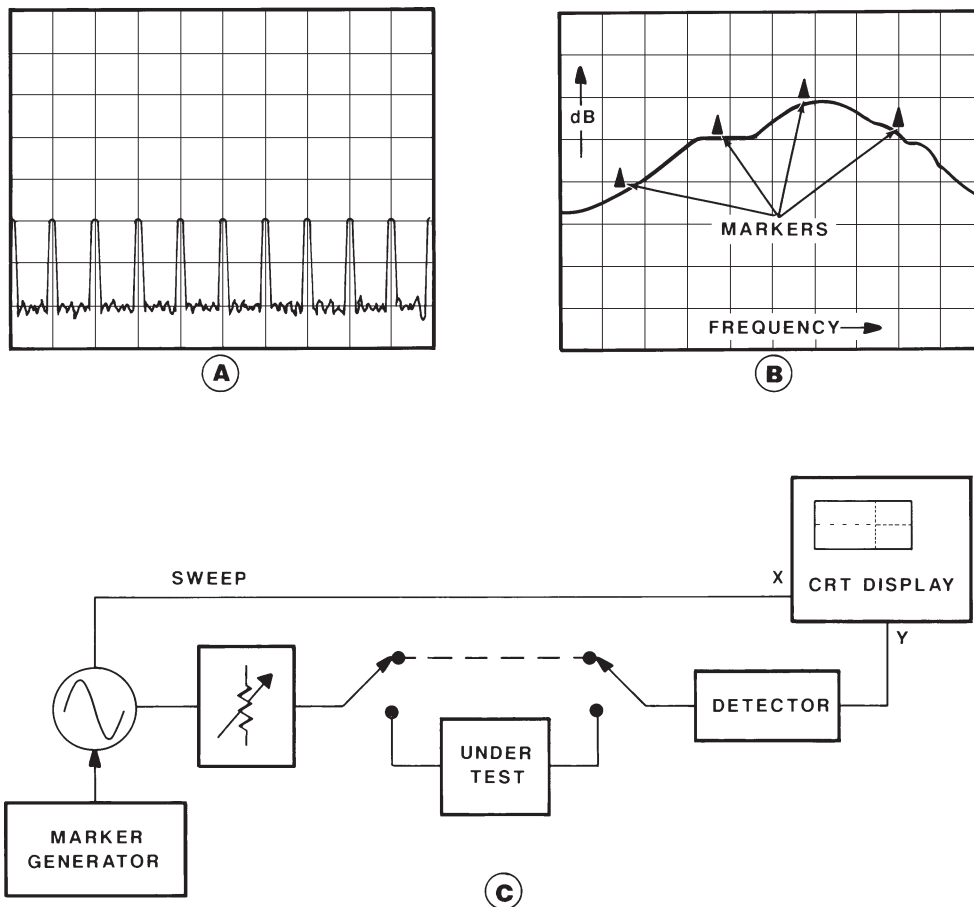


Figure 8-2

Output signals from comb and marker generators.

Figure 8-3A shows the frequency distribution of a standard multiplex arrangement of three channels at 4 kHz increments of 12, 16, and 20 kHz. The fundamental frequency of the harmonic generator producing the 12, 16, and 20 kHz signals is 4 kHz. The passband of a filter at the output of the harmonic generator is from 10 to 22 kHz, rejecting all other harmonics from the circuit. Beyond this first filter, the signals pass through an equalizing network to bring each of the three harmonics to the same power level. Finally, each of the individual harmonics are passed through a narrow-band filter to produce sine waves at their respective frequency. The filters also isolate each signal frequency (12, 16, and 20 kHz) from the others in the group.

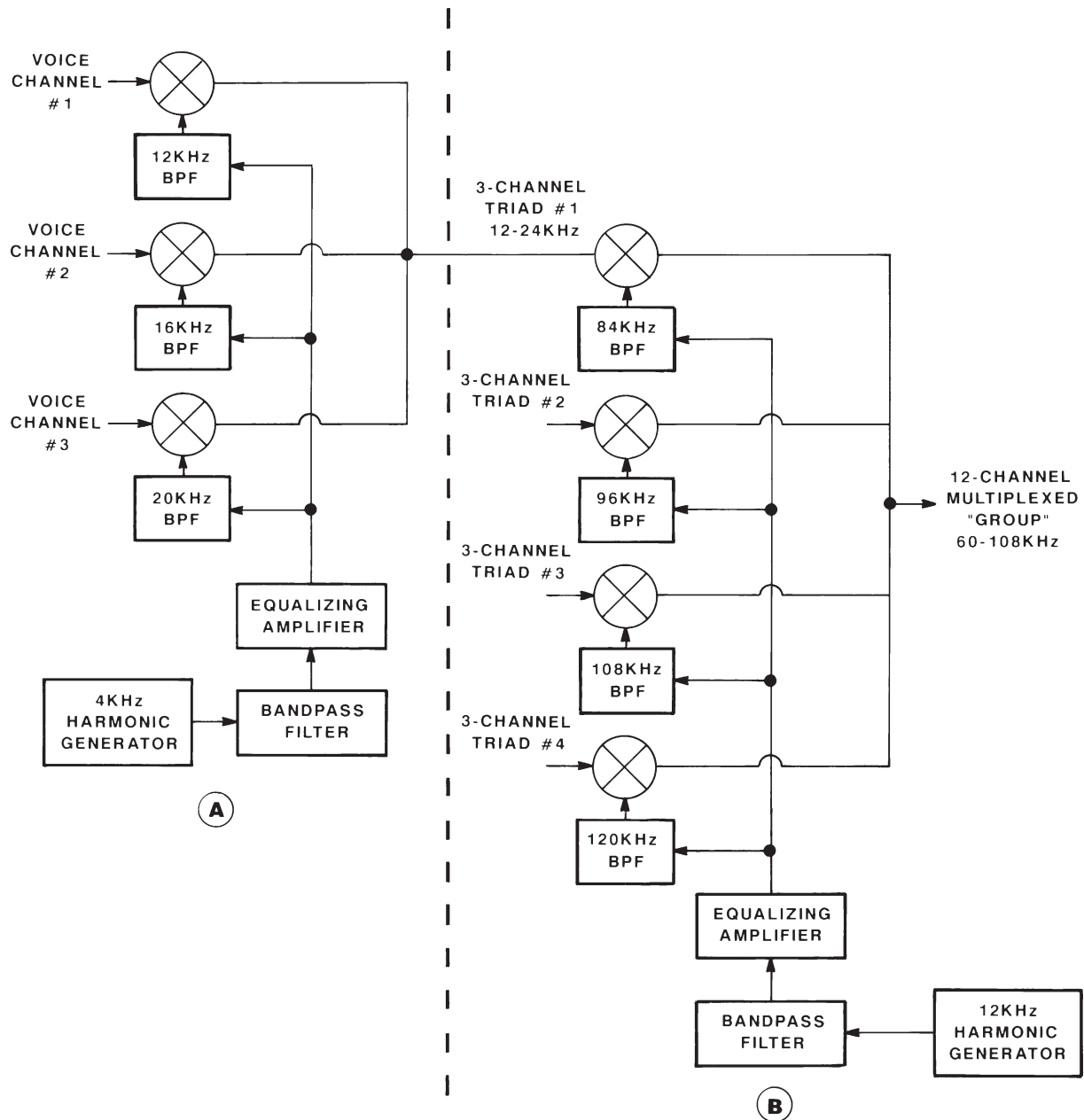


Figure 8-3

Frequency division multiplex of 12-channel group.

Figure 8-3B shows how the multiplexed output signal triad from Figure 8-3A, and three other, three-channel multiplexers are combined. The frequency increments of this second stage in the multiplexing process is 12 kHz. The 84, 96, 108, and 120 kHz signals are generated as the seventh through tenth harmonics of the very stable 12 kHz fundamental frequency generator. The passband of the filter at the output of this harmonic generator is from 90 to 126 kHz. The signals are equalized to the same power level, then filtered, as with the lower frequency section of the multiplex circuit.

HARMONIC MULTIPLIERS

There are many circuits used in communications systems that utilize one or more harmonic frequencies of signals applied to their inputs. These circuits are identified by such names as “doubblers,” “triplers,” or “multipliers.” Such multipliers are specified by the range of signal frequencies acceptable at their inputs, and the factor by which the signal frequencies will be multiplied.

Multipliers are used to obtain higher-frequency signals that proportionally track, and are synchronized to lower-frequency signals. Multipliers are also used to generate stable VHF, UHF, SHF, and EHF radio signals. Almost all FM radio broadcast transmitters, for example, originate their station frequencies at stable lower frequencies, before frequency multiplying up to their 88 to 108 MHz broadcast frequency.

Harmonic multipliers create higher frequency signals that are harmonics of a lower-frequency reference signal. The higher-frequency output signal will track, and retain the properties of the multiplied lower-frequency input signal.

A 10–20 kHz doubler, for example, will multiply that range of signal frequencies applied to its input, by a factor of two. The signals at its output will range from 20 to 40 kHz, twice any input signal in the range of 10 to 20 kHz. If the same range of input frequencies are applied to a frequency tripler, its output signals will extend from 30 to 60 kHz. Figure 8-4 shows examples of the doubler and tripler described in these paragraphs.

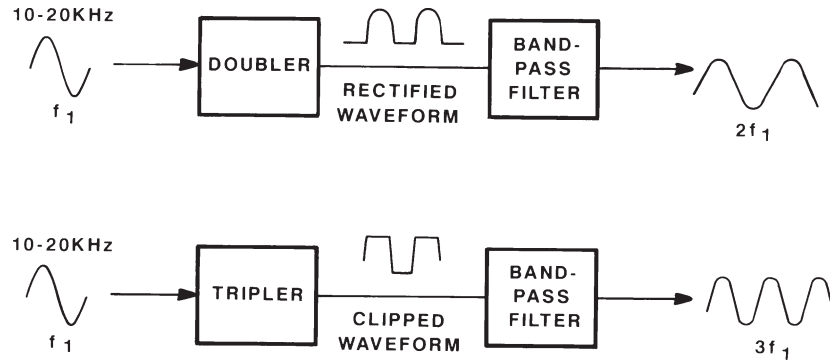


Figure 8-4

Frequency doubler and tripler.

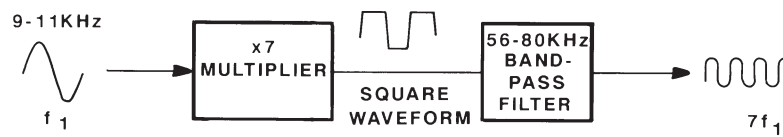
Doublers and triplers operate with an input bandpass frequency response at, or less than, an octave. With response just slightly below an octave, all harmonics will be suppressed at the multiplier's inputs. (The second harmonic of any signal is an octave higher than that signal's fundamental frequency.) For the doubler of our example, the input frequency limits of 10 to 20 kHz will suppress all harmonics from the input (except its unlikely operation at exactly 10.000 kHz) to assure its sine waveshape.

A well designed multiplier will distort its input sinewave signal in a way that produces the optimum power level for the harmonic number it is designed to generate. A doubler will rectify the sine waveform to obtain a higher-level second harmonic, and much weaker third harmonic signal at its output. A tripler will symmetrically clip the sine waveform to obtain a higher-level third harmonic, and much weaker second and fourth harmonic signals at its output. (Refer back to Figure 5-3 on page 5-9 to review the harmonic structure of these waveshapes.)

Low-distortion sinewaves are almost always required at the outputs of frequency multipliers. Filters are used at the output of the multipliers to assure sine waveforms. The output filters are tuned to the frequency multiple specified for the multiplier ($\times 2$, $\times 3$, $\times 7$, etc.). The frequency multipliers with

higher multiplying factors require proportionally narrower bandpass output filters to select only one of the output harmonic signals.

For example, input signal limits from 10 to 20 kHz will provide extended harmonic signals at the output of the $\times 7$ multiplier of Figure 8-5. With 10 kHz at its input, it will have 70 kHz at its output. However, other harmonics will also be present at the output. (Remember, a frequency multiplier is a harmonic generator!) If the multiplier clips its input signal waveform, there will be significant odd numbered harmonics at the output of the multiplier; including frequencies at 30, 50, 70, 90, 110 kHz, and so-on. The multiplier will not multiply $\times 7$ exclusively.



f_1 FREQ.	HARMONIC NUMBER		
	$\times 5$	$\times 7$	$\times 9$
9KHz	45KHz	63KHz	81KHz
10KHz	50KHz	70KHz	90KHz
11KHz	55KHz	77KHz	99KHz

Figure 8-5

Frequency multiplier and output filtering.

To make this component a true $\times 7$ multiplier, it must have a frequency-selective filter at its output. This filter must exclude all frequencies other than the $\times 7$ harmonic multiple. Bandpass limits must be set into the output filter. Unfortunately, the reduced output passband will also limit the input passband of the multiplier. Higher multiplier factors require a limited range of input frequencies.

As an example, the Table of Figure 8-5 shows the fifth, seventh, and ninth harmonics of three frequencies, with 10 kHz at the center frequency. Looking at this table, assume that you want to operate the $\times 7$ multiplier from 9 to 11 kHz at its input. To do so, you must exclude the fifth harmonic of the higher frequency (11 kHz $\times$ 5 = 55 kHz), and the ninth harmonic of the lower frequency (9 kHz $\times$ 9 = 81 kHz) input signals. Therefore, the multiplier's output filter must be limited to a passband from above 55 kHz to below 81 kHz.

The trade-offs for frequency multiplication include a reduction in bandpass and frequency range, as well as a reduced output conversion efficiency. Conversion efficiency, expressed as a percentage or in dB, is simply the ratio of the power level of the specific harmonic signal, compared to the fundamental-frequency signal. The higher-factor multipliers typically have harmonics at much lower power levels than the fundamental frequency they multiply, so their conversion efficiency is poor. Typical $\times 7$ multipliers, for example, may have output signals at only 1% to 0.1% (–20 dB to –30 dB) of their input signal levels.

You may calculate the conversion efficiency (Eff_{conv}) of a harmonic multiplier, based upon known or measured power levels of the fundamental (P_{fund}) and harmonic (P_{har}) signals, as follows:

$$\% \text{Eff}_{\text{conv}} = \left(\frac{P_{\text{har}}}{P_{\text{fund}}} \right) \times 100$$

$$\% \text{Eff}_{\text{conv}} = 10 \log \left(\frac{P_{\text{har}}}{P_{\text{fund}}} \right)$$

For example, a 1-watt signal applied to a $\times 5$ multiplier yields an output power level of 4 milliwatts at five times the input frequency. The conversion efficiency of the multiplier is calculated as:

$$\% \text{Eff}_{\text{conv}} = \left(\frac{P_{\text{har}}}{P_{\text{fund}}} \right) \times 100$$

$$\% \text{Eff}_{\text{conv}} = \left(\frac{0.004 \text{ W}}{1} \right) \times 100$$

$$\% \text{Eff}_{\text{conv}} = 0.4\%$$

$$\% \text{Eff}_{\text{conv}} = 10 \log \left(\frac{P_{\text{har}}}{P_{\text{fund}}} \right)$$

$$\% \text{Eff}_{\text{conv}} = 10 \log \left(\frac{0.004 \text{ W}}{1 \text{ W}} \right)$$

$$\text{dB Eff}_{\text{conv}} = -24 \text{ dB}$$

SUMMARY

A wideband circuit is required to transmit a complex waveform with its significant harmonics. The primary advantage of using a wideband circuit is the accuracy with which it passes signals. With this advantage comes several disadvantages. Wideband circuits limit the channel capacity of a system, have higher noise power, consume more power, and induce phase distortion.

A narrow-band circuit, on the other hand, uses filters to suppress the higher-frequency harmonics. This reduces the circuit's ability to precisely reproduce the signal, but reliable communication is still achieved. Telephone communications use narrow-band circuits.

Many signal processing functions used in communications systems rely on the presence of harmonic signals. Harmonic generators are used to add harmonics to an existing waveform. They are also used to create signals equally in frequency, and to produce the carrier frequencies used in a frequency division multiplex system. A non-linear circuit creates harmonics by distorting the signals applied to its input. Linear circuits, on the other hand, output the exact same waveform that is applied to its input.

SELF-TEST REVIEW

1. What is the primary advantage of wideband circuits over narrow-band circuits in communications systems?
 - A. Accuracy.
 - B. Channel capacity.
 - C. Lower noise level.
 - D. Power efficiency.
2. A narrow-band filter is tuned to the fundamental frequency of a square-wave signal passing through it. What is the waveform at the filter's output?
 - A. Square wave.
 - B. Sine wave.
 - C. Triangle wave.
 - D. Pulse wave.

3. Why do operating restrictions require the output amplifiers of radio transmitters to be heavily filtered?

4. A signal waveform is “distorted” when passing through a circuit, in a way that increases its harmonics. What is the correct name for this type of distortion? _____
5. A perfect sinewave signal is processed through a non-linear circuit. What signal cannot appear at the circuit’s output?
 - A. A perfect sine wave.
 - B. A square wave.
 - C. A triangle wave.
 - D. A rectified sine wave.
6. It is desired to obtain the third harmonic of a signal at the output of a circuit, with only a minimum of filtering to suppress the signal’s second and fourth harmonics. Which waveform will be the best choice for this process?
 - A. Rectified sine.
 - B. Triangle.
 - C. Square.
 - D. Pulse.
7. If it is required to retain the waveform of audio signals up through their fifth harmonic, what is the highest fundamental signal frequency that can be reproduced accurately through a circuit that is low-pass filtered at 5 kHz? _____
8. What is the incremental spacing of frequencies in a comb generator if its fundamental is 10 kHz? _____
9. A marker generator displays marks at 100, 105, 110, 115, 120, and 125 kHz on the face of a network analyzer’s CRT display. What is the fundamental frequency of the marker generator? _____

10. What is the advantage of frequency multipliers and harmonic generation of signals, for multiplexing voice channels as shown in Figure 8-3 on page 8-10, rather than directly generating each frequency?
-
-
11. What is the ideal waveform for a frequency doubler?
- A. Rectified sine.
 - B. Triangle.
 - C. Square.
 - D. Pulse.
12. What is the ideal waveform for a frequency tripler?
- A. Rectified sine.
 - B. Triangle.
 - C. Square.
 - D. Pulse.
13. Which harmonic numbers must be considered in determining output filtering and input frequency limits for a $\times 7$ multiplier?
- A. sixth and eighth.
 - B. fifth and sixth.
 - C. eighth and ninth.
 - D. fifth and ninth.
14. A pulse waveform similar to the one shown in Figure 5-3B, on page 5-9, is created to be used in a frequency-multiplier circuit. Referring to Table 8-1, on page 8-7, which multiplier factor will result in the best conversion efficiency?
- A. $\times 3$
 - B. $\times 4$
 - C. $\times 5$
 - D. $\times 6$

15. Refer to Table 8-1, on page 8-7, and Figure 5-3, on page 5-9, to evaluate the harmonic structure of the sawtooth waveform; then, graph the frequency response curve, in Figure 8-6, for an equalizing amplifier to bring the second through the tenth harmonic of this waveform to equal power levels.

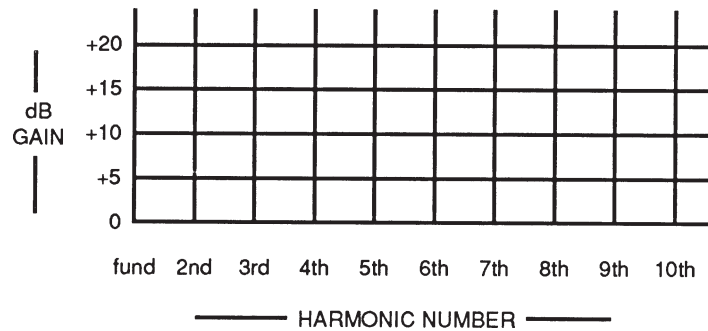


Figure 8-6

Figure for question 15.

16. A 10 kHz signal is to be processed through a $\times 5$ frequency multiplier, with clipping distortion that suppresses the fourth and sixth harmonics. The input frequency range must extend from 9 kHz through 11 kHz. What are the output filter's frequency limits, as required to suppress the third and seventh harmonics of the input frequency limits of 9 kHz through 11 kHz? _____
17. Refer to Table 8-1, on page 8-7, and Figure 5-3E, on page 5-9, to determine the characteristics of a frequency doubler that rectifies the sinewave signal applied to its input, then calculate the conversion efficiency of the doubler based upon this data. _____ dB

Section 3

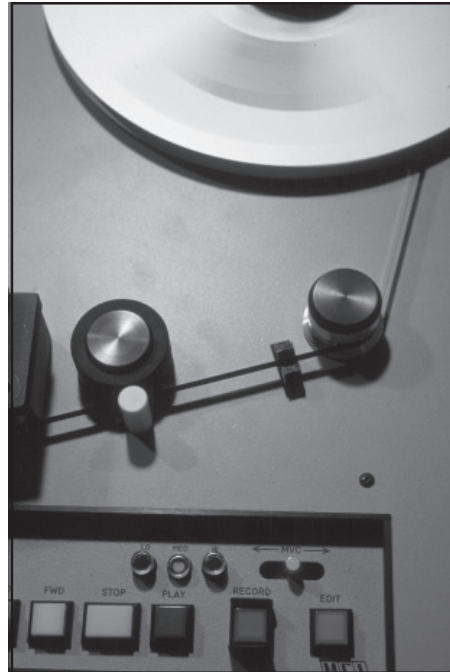
COMBINING SIGNALS

CHAPTER 9 — Multiple Signals in Linear Circuits

INTRODUCTION

There are many circuits in telecommunication systems that simultaneously process more than one signal. In some signal processing functions, two or more signals must be carried on a single pair of wires, without their interfering with each other. One of the more common examples of this process occurs with telephone systems, where transmit (talk) and receive (listen) signals don't interfere as they pass along the same circuit. The signals affect the total circuit current, but they don't affect each other's waveforms.

Another example of combining many signals into a single circuit, while maintaining their distinctions, is music recording. In a recording studio, a number of microphones are placed at different locations among the performing musicians. The output signal from each microphone is connected to a mixing console. The mixing is done by control-board circuits that combine the microphone signals. If the music is to be accurately reproduced, the recording engineer at the console adjusts only the *level* of the respective microphone signals, without altering their waveforms.



Each of the multiple signals within a circuit produce their own distinct waveform. When these waveforms are combined into a single circuit, the resulting combined waveform will include all of these distinct signals. Your ability to interpret these combined waveforms will allow you to determine the character of each of the individual signals that the waveforms contain. This kind of analysis uses the procedures of the superposition theorem.

In this chapter, we will describe the results and purposes of processing multiple signals through linear filter circuits. When two (or more) signals are processed through linear circuits, the signals remain distinct, and they retain their initial waveform and harmonic structure. The signals' instantaneous voltages merely add to produce a composite waveform.

OBJECTIVES

When you complete this chapter, you will be able to:

- Identify and distinguish between linear and non-linear circuits that process two (or more) signals.
- Calculate the effects of linear mixing (adding or summing) of two signals through a single circuit using superposition techniques.
- Determine and identify the waveforms that result from the adding, or linear mixing, of signals.

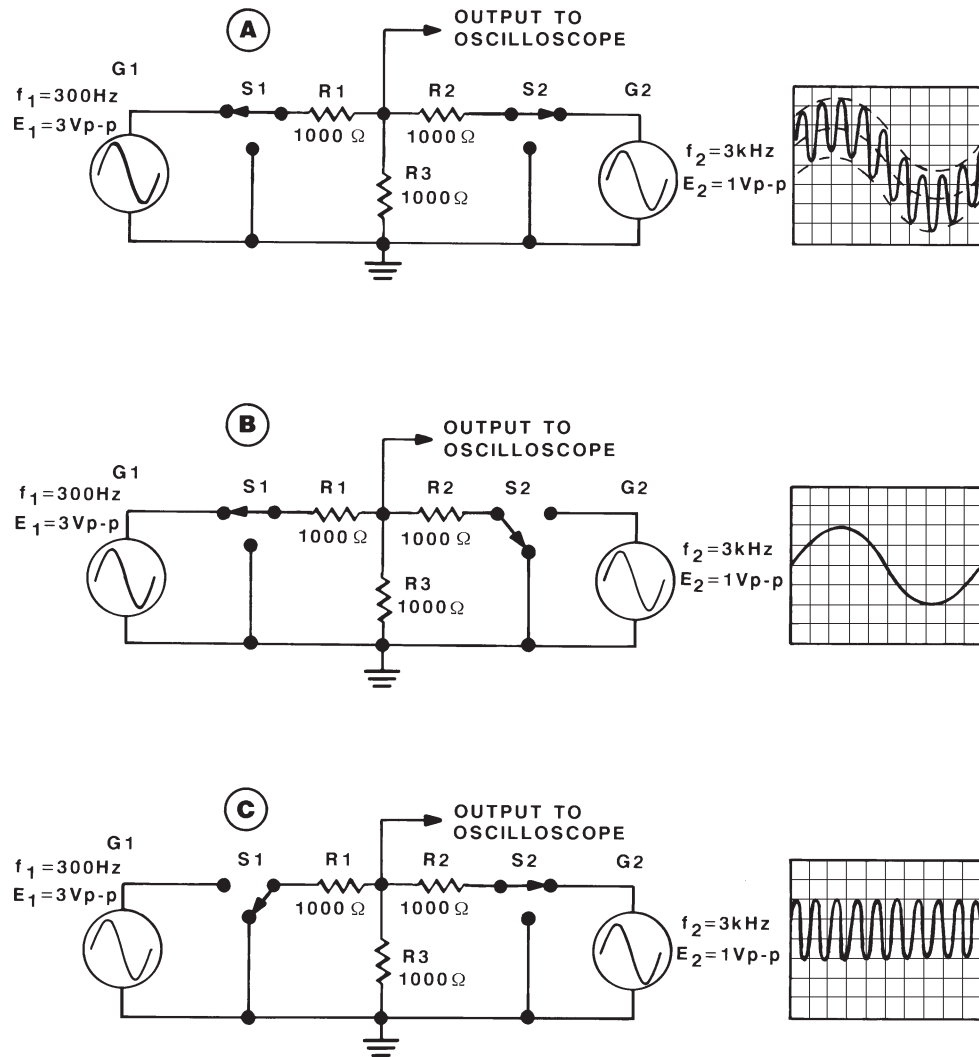
SUPERPOSITION

You may recall the superposition theorem from your study of electrical fundamentals. The theorem simply states that “the whole is the sum of its parts.” Superposition is used to determine the currents and voltages within an electrical network, when there is more than one source of power applied to the network.

The term *superposition* evolved from “superimpose.” With two sources of power, one source is *superimposed* upon the other across an electrical network. The resulting voltages and currents within the network are simply the algebraic sum of the individual voltages and currents from each source. The calculations are straight forward for DC power sources. Calculating the voltages and currents for AC and signal sources is somewhat more complicated, as the signals are changing in instantaneous phase and amplitude (level). However, each of the signals will maintain their distinct characteristics as they pass through circuits. So the effects of each signal can be individually determined, and then combined.

Superposition is the word used to identify the superimposing of one source of power upon another in a single circuit. The resulting total power is the algebraic sum of the individual power sources.

Figure 9-1 shows an example of a circuit that has two signal sources applied to each side of a “T-network.” If both switches S1 and S2 connect the generators to the circuit, as in Figure 9-1A, both signals will primarily affect the voltage and current of resistor R3. The effect of generator G1 alone can be determined by switching S2 to remove G2 from the circuit, as shown in Figure 9-1B. And the effect of G2 alone can be determined by connecting G2 to the circuit, and removing G1 from the circuit, as shown in Figure 9-1C. The resulting waveform on the oscilloscope display is simply the 3 kHz signal of G2, “riding on” (superimposed) the 300 Hz signal of G1.

**Figure 9-1**

Combining signals in a linear circuit.

Superposition Calculations

The combined signal waveform of Figure 9-1A is typical of those found in telephone circuits. The higher-level signal (G1) can represent a transmit signal, while the lower-level signal (G2) can represent the receive signal. (The frequencies for this example, 300 Hz and 3 kHz, are extreme for illustrative purposes.) The values for each signal voltage across R3 can be determined using basic network calculations for each signal source.

The first calculation assumes G1 is connected to the circuit, and G2 is removed from the circuit, per Figure 9-1B. For G1:

NOTE: This symbol $\parallel$ means parallel to.

$$E_{R3} = E_{G1} \left(\frac{R2 \parallel R3}{R1 + (R2 \parallel R3)} \right)$$

$$R2 \parallel R3 = \left(\frac{R1 \times R3}{R2 + R3} \right)$$

$$R2 \parallel R3 = \left(\frac{1k \times 1k}{1k + 1k} \right)$$

$$E_{R3} = 3 \left(\frac{500}{1k + 500} \right) = 1 \text{ V}_{p-p}$$

$$E_{R3} = 0.5 \text{ V peak}$$

Like the first calculation, the second calculation for G2 assumes that G2 is connected to the circuit, and G1 is removed from the circuit, per Figure 9-1C. For G2:

$$E_{R3} = E_{G2} \left(\frac{R1 \parallel R3}{R2 + R1 \parallel R3} \right)$$

$$R1 \parallel R3 = \left(\frac{R1 \times R3}{R1 + R3} \right)$$

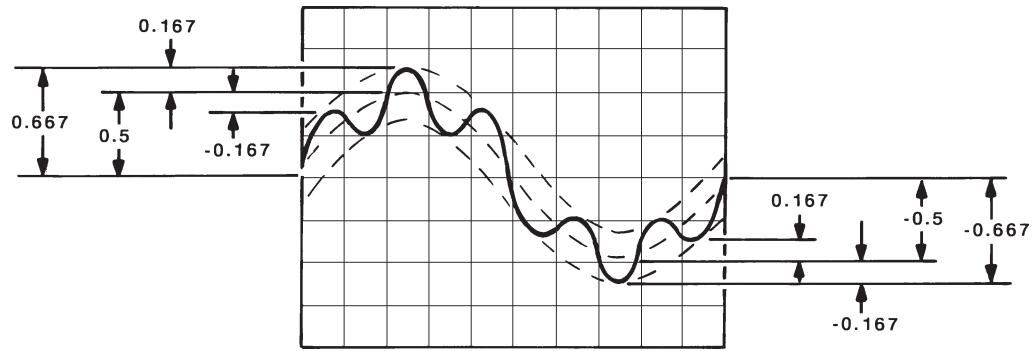
$$R1 \parallel R3 = \left(\frac{1k \times 1k}{1k + 1k} \right)$$

$$R1 \parallel R3 = 500 \text{ ohms}$$

$$E_{R3} = 1 \left(\frac{500}{1k + 500} \right) = 0.167 \text{ V}_{p-p}$$

$$E_{R3} = 0.17 \text{ V peak}$$

The two signal voltages are applied simultaneously to R3, rising to their maximum instantaneous peak values. Therefore, the signal from the first generator (G1) will rise to ± 0.5 volts; while the signal from the second generator will rise to ± 0.167 volts. The voltage across R3 will be the algebraic sum of the instantaneous voltages from the sources, as scaled in Figure 9-2.

**Figure 9-2**

Superposition of signal voltages.

At the instant when both signals are in their positive phase, the voltage across R3 will be:

$$E_{R3} + E_{R3}' = 0.5 + 0.167$$

$$E_{R3} + E_{R3}' = 0.667 \text{ V peak}$$

At an instant later, when the signal from G1 is +0.5 V and the signal from G2 is -0.167 V, the voltage across R3 will be:

$$E_{R3} + E_{R3}' = 0.5 + (-0.167)$$

$$E_{R3} + E_{R3}' = 0.33 \text{ V peak}$$

Later, when both signals are in their negative phase, the voltage across R3 will be:

$$E_{R3} + E_{R3}' = (-0.5) + (-0.167)$$

$$E_{R3} + E_{R3}' = -0.667 \text{ V peak}$$

Finally, at an instant later, when the signal from G1 is -0.5 V and the signal from G2 is +0.167 V, the voltage across R3 will be:

$$E_{R3} + E_{R3}' = (-0.5) + (0.167)$$

$$E_{R3} + E_{R3}' = -0.33 \text{ V peak}$$

Calculating Voltage and Frequency

The waveform of Figure 9-2 is typical of those found in linear circuits. Each of the signals are distinct, and appear on an oscilloscope as a higher-frequency signal *riding* upon a lower-frequency signal. This figure illustrates the combination of only two signals to keep the concepts simple. With experience, you should easily identify multiple signals within the combined waveform of the oscilloscope display; and then, determine the properties of the individual signals within the waveform by using superposition techniques.

If you wish to determine the properties of the higher frequency signal in a waveform such as in Figure 9-3, for example, it can be done in two steps. First, the peak signal voltage can be calculated as the difference average of the maximum and minimum *high-frequency* portion of the waveform. This is usually measured at the peak of the *lower-frequency* signal's signal swing, point "A" in Figure 9-3, and referenced to the zero crossing point of the waveform.

$$E_{f-hi}(\text{peak}) = \frac{E_{\max} - E_{\min}}{2}$$

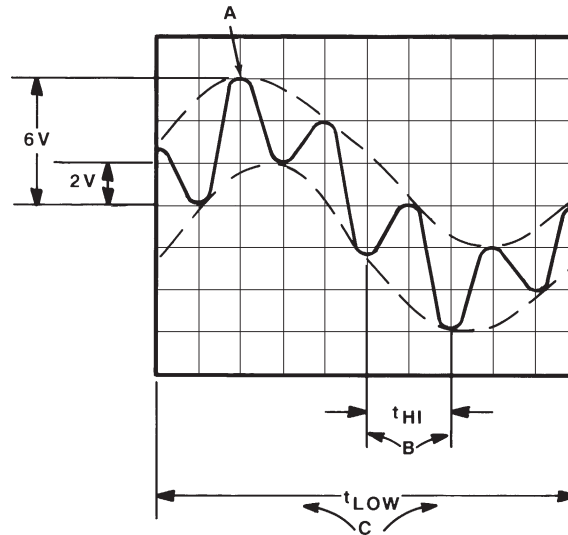


Figure 9-3
Waveform voltage and time dimensions.

With the maximum voltage at 6 volts, and the minimum voltage at 2 volts, in the waveform, the peak voltage of the higher-frequency signal is:

$$E_{f\text{-hi}}(\text{peak}) = \frac{6 \text{ V} - 2 \text{ V}}{2}$$

$$E_{f\text{-hi}}(\text{peak}) = 2 \text{ V peak}$$

Second, the frequency of the higher-frequency signal is determined as the reciprocal of its time-period. With the period of time between peaks in the waveform equal to 0.5 milliseconds, "B" in Figure 9-3, the frequency of the signal is:

$$f_{\text{hi}} = \frac{1}{t}$$

$$f_{\text{hi}} = \frac{1}{0.5 \times 10^{-3}}$$

$$f_{\text{hi}} = 2000 \text{ Hz}$$

The properties of the lower-frequency signal in the waveform of Figure 9-3 can also be determined in two steps. First, the peak signal voltage is calculated as the sum-average of the maximum and minimum *high-frequency* portion of the waveform. The calculation is made at the peak of the *lower-frequency* signal's swing, point "A" in Figure 9-3.

$$E_{f\text{-low}}(\text{peak}) = \frac{E_{\text{max}} + E_{\text{min}}}{2}$$

The maximum voltage is again 6 volts, and the minimum voltage is 2 volts. In the waveform, the peak voltage of the lower-frequency signal is:

$$E_{f\text{-low}}(\text{peak}) = \frac{6 \text{ V} + 2 \text{ V}}{2}$$

$$= 4 \text{ V peak}$$

Then, the signal frequency can be determined as the reciprocal of the time-period of the lower-frequency signal. With the period between peaks in the waveform at “C” in Figure 9-3 at 4 msec., the frequency of the signal is:

$$f_{\text{low}} = \frac{1}{t_{\text{low}}}$$

$$f_{\text{low}} = \frac{1}{4 \times 10^{-3}}$$

$$f_{\text{low}} = 250 \text{ Hz}$$

LINEAR CIRCUITS

The electrical network in Figure 9-1 is a linear circuit. The output voltage and current at resistor R3 is *directly proportional* to the applied signal voltages and currents. The effects of superposition, the direct adding of one signal to another, can occur only in a linear circuit. Any circuit non-linearity will result in signal distortions that complicate the direct addition of the signals.

The adding of one signal to another, superposition, can occur only in a linear circuit.

Purely resistive and/or reactive circuits, circuits which contain only resistors, capacitors, and inductors, are linear circuits because they don't have components that distort the waveshape of the signals applied to them. Even if the applied signals are not sine waves, no *additional* frequencies (or harmonics) will be created as a result of the signal passing through a linear circuit. If the circuits contain LC filter elements, they will *remove* harmonics from complex (non-sinewave) waveforms, but they *won't create* other signal frequencies.

Linear circuits don't distort signal waveforms and they don't create additional frequencies.

With electronic devices, such as transistors, IC's, and diodes, there is a trade-off for linearity vs. gain and/or output power. Most electronic devices are linear within some limits of their operation, but they will distort signals if these limits are exceeded. Where gain is required, electronic circuits must be carefully designed to assure their linearity. Also, the signal levels applied to these circuits must be critically set or calibrated to obtain the best circuit performance.

Electronic amplifying circuits that are especially designed for linear operation are aptly called "linear amplifiers." Those circuits that are specifically intended to process *multiple* signals are identified as "combiners," "summing amplifiers," and "adders." In all such circuits, the limits of their operation are clearly specified to assure their best linearity.

The people who work with signal processing circuits and signals in communications equipment understand linearity, and they are skilled in adjusting circuit operation and signal levels. To measure circuit linearity, they work backwards; they measure non-linearity. Circuit non-linearity results in signal distortion; so, the circuits that produce the *least* distortion are the most linear. To evaluate *linearity*, you must also thoroughly understand its reverse property, *non-linearity*. The details of non-linearity and signal distortion will be described in the next chapter.

SUMMARY

In this chapter, we reviewed how two or more signals interact with each other as they are processed through linear electrical and electronic circuits. With linear circuits, the signals remain distinct in character, even though they pass along the same circuit paths. The waveforms in these signals are merely the instantaneous sum of the voltages in each of the signal's waveforms. The mathematical theorem used to calculate the instantaneous voltages in these waveforms is called "superposition." The frequency and voltage level of each of the linearly combined signals can be obtained by examining the waveforms displayed on an oscilloscope.

SELF-TEST REVIEW

1. When two input signals are combined in a linear circuit, the resulting output waveform contains:
 - A. Both signals and their harmonics.
 - B. Both signals, and frequencies in addition to their harmonics.
 - C. Both signals only.
 - D. Harmonics and additional frequencies only.

2. In a simple DC network, two sources provide +4 and +6 volts individually across a resistor common to the currents from both sources. What is the voltage across the resistor when both sources are applied simultaneously?
 - A. 1.5 VDC.
 - B. 2 VDC.
 - C. 10 VDC.
 - D. 24 VDC.

3. What is the name of the technique used to solve electrical network problems such as described in Question 2? _____

4. Two signals are applied across the T-network of Figure 9-4. What is the network voltage if the *higher-frequency source is at its peak*, and the lower-frequency wave is crossing zero (changing phase from + to -)?
 - A. 0 volts.
 - B. 0.167 volts.
 - C. 0.33 volts.
 - D. 0.5 volts.

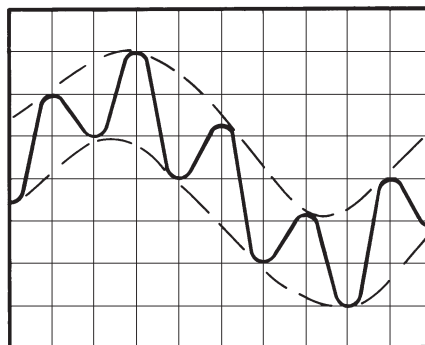
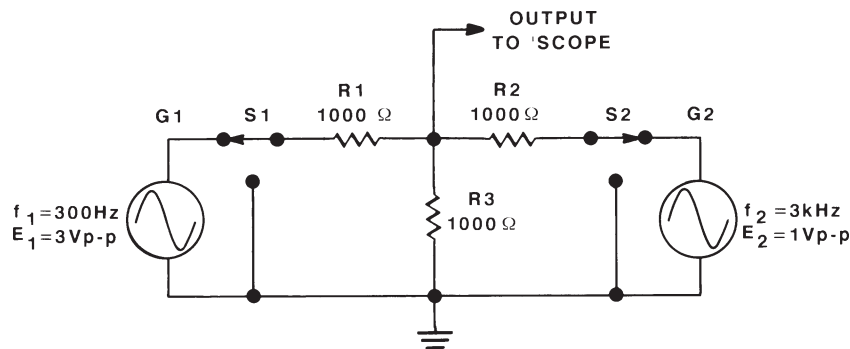

Figure 9-4

Figure for Questions 4, 5, 6, and 7.

5. Draw an arrow in Figure 9-4 that points to the location on the waveform described in Question 4. Mark this arrow with the number 5.
6. Two signals are applied across the T-network of Figure 9-4. What is the network voltage if the *lower-frequency source is at its peak*, and the higher-frequency wave is crossing zero (changing phase from + to -)?
 - A. 0 volts.
 - B. 0.33 volts.
 - C. 0.5 volts.
 - D. 0.667 volts.
7. Draw an arrow in Figure 9-4 that points to the location on the waveform described in Question 6. Mark this arrow with the number 7.
8. An electronic amplifier does *not* distort the waveforms of the signals applied to its input. The amplifier is classified as:
 - A. Linear.
 - B. Non-linear.
 - C. Resistive.
 - D. Reactive.
9. Apply the techniques of superposition to calculate the values indicated for the T-network in Figure 9-5.
 - A. E_{R3} for generator G1. _____
 - B. $E_{R3'}$ for generator G2. _____
 - C. The *peak* instantaneous voltage across R3 when both of the signals are in phase and at their positive peak values.
 - D. The *peak* instantaneous voltage across R3 when both signals are applied; with G1 at its negative peak, and G2 is at its positive peak.

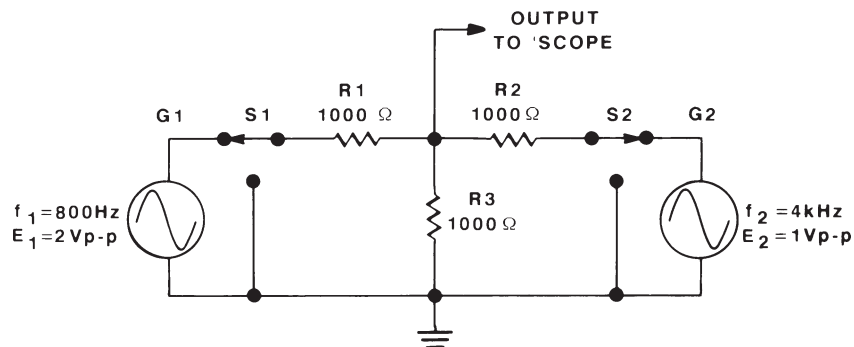


Figure 9-5

Figure for Question 9.

10. Refer to Figure 9-6, and calculate the values indicated:

- A. The frequency of signal f1. _____
- B. The frequency of signal f2. _____
- C. The peak voltage of signal f1. _____
- D. The peak voltage of signal f2. _____

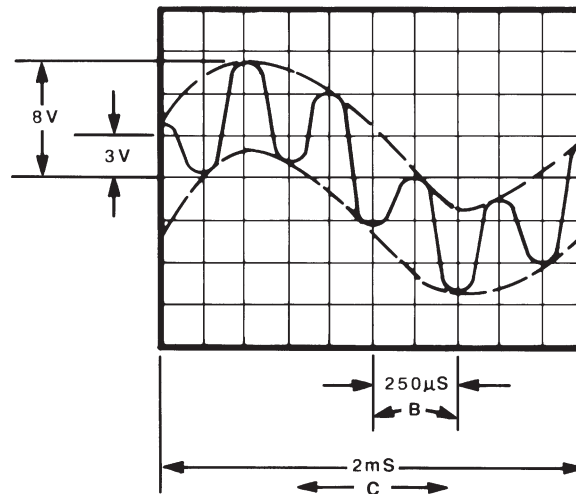


Figure 9-6

Figure for Question 10.

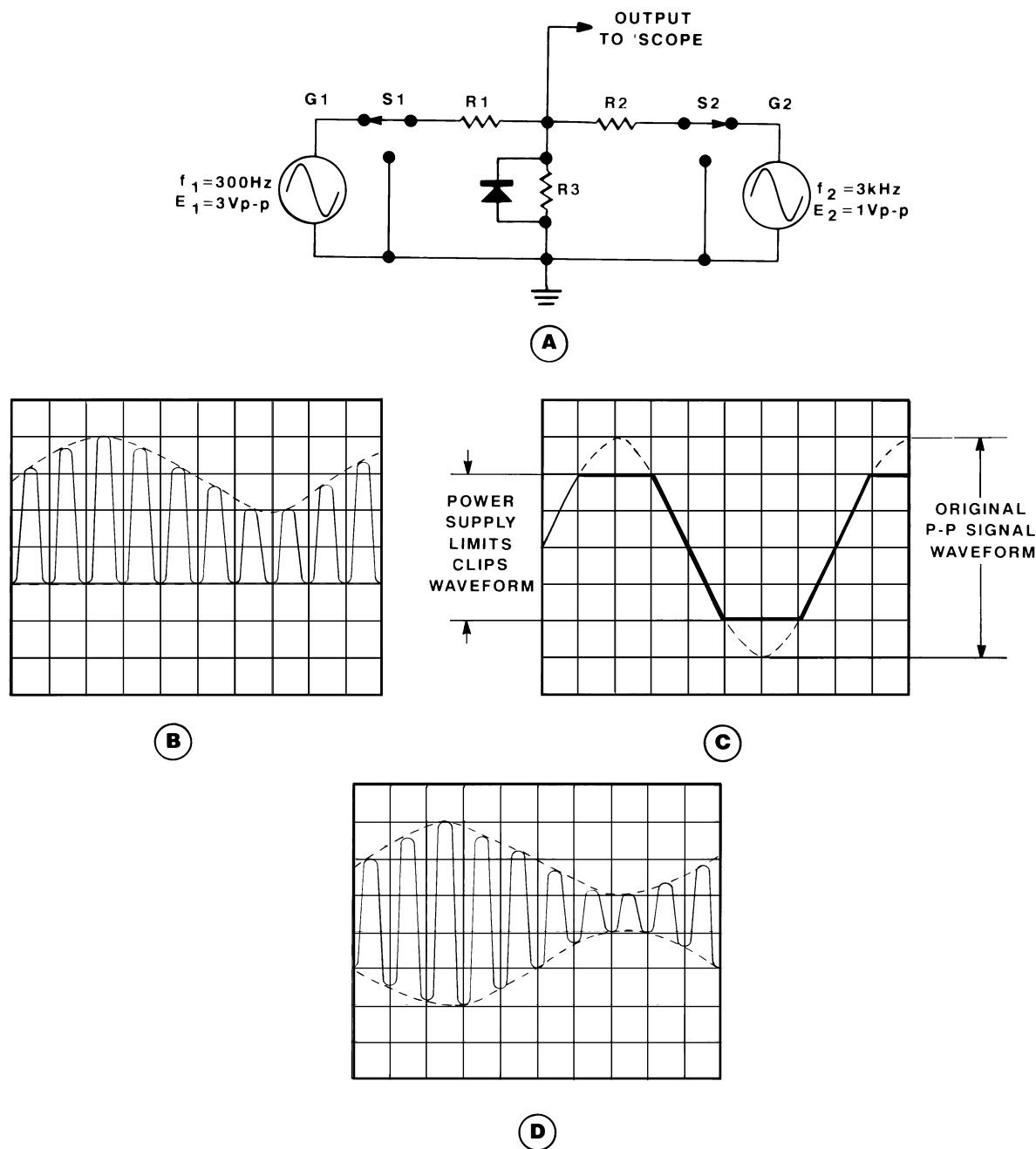
CHAPTER 10 — Multiple Signals in Non-Linear Circuits

INTRODUCTION

In previous chapters, we described how electronic circuits caused waveform distortion which added harmonics to signals they processed. Just as distortion occurs with one signal, it also occurs when multiple signals are processed through circuits. With multiple signals, however, the distortion produces other frequencies in addition to harmonics. These other frequencies result from the applied signals themselves interacting with each other. The frequencies that occur as a result of this multi-signal interaction are called “intermodulation products” or “intermodulation frequencies.” As with harmonics, there are some applications where intermod products are undesirable. These frequencies are classified as “intermodulation *distortion*.” More often, however, intermod signals are purposely created in communications equipment, where they are *not* defined as distortion.

Intermodulation products are non-harmonic signals at frequencies *other* than those *applied* to a non-linear circuit, which are created as a result of the signals interacting with each other within the circuit.

A circuit that is *perfectly linear* will not distort, but signal distortion will occur if it is non-linear. If two (or more) signals are applied to a non-linear circuit, it will produce both harmonic and intermodulation frequencies as a result of waveform distortion. So *both* harmonics *and* intermod products can exist together, even when *only* intermod frequencies are desired. The suppression of these harmonics will often improve the performance of the non-linear circuits. The simple linear network of Figure 9-1 (on page 9-4) is purely resistive, and will not distort the signals passing through it. If a diode is placed across R3, however, the circuit will become non-linear. Figure 10-1A, on the next page, shows the same simple T-network with a diode across the common signal path through R3. The diode distorts the waveforms of both signals, as shown in Figure 10-1B, causes the circuit to become non-linear, and produces harmonics and intermodulation frequencies.

**Figure 10-1**

Combining signals in a non-linear circuit.

Like the diode, electronic amplifiers and signal processing circuits will distort waveforms. One common type of distortion results from signals exceeding the power supply limits of transistors and ICs. A high signal level applied to an amplifier will cause the signal waveform to clip at its peaks, and become distorted, as shown in Figure 10-1C. The resulting waveform is shown in Figure 10-1D. (Compare this waveform with the linear displays of Figures 9-2 and 9-3, in the previous chapter.)

OBJECTIVES

When you complete this chapter, you will be able to:

- Determine and identify the waveforms that will result from the processing of two signals through non-linear circuits.
- Determine and identify the intermodulation frequencies that result from processing two signals through non-linear circuits.
- Calculate the percentage factors of intermodulation and harmonic distortion in circuits and signals.
- Calculate the power distribution of intermodulation signals over their range of frequencies.

INTERMODULATION DISTORTION

As mentioned earlier, a different type of distortion (in addition to harmonic distortion) occurs when two or more signals are applied to a non-linear circuit. This type of distortion is called “intermodulation distortion” (“intermod,” for short). Intermod, is a measure of how multiple signals affect each other, and create intermod frequencies (or intermod products) as they pass through electronic circuits.

There is a difference between the signals created as a result of intermodulation distortion and those created by harmonic distortion. The difference exists in both the *number and frequencies* of the distortion signals, and in the ways by which the distortion is created. Consequently, harmonic and intermod distortion specifications may often differ for the same circuit. We’ll assume that you understand harmonics from your study of Chapter 5, so let’s proceed to examine how two signals interact to produce intermodulation frequencies.

Signals created as a result of intermod distortion differ in both the *number and frequency* from those signals created as a result of harmonic distortion.

Figure 10-2 shows the non-linear combining of two sine waves of different frequencies. The unusual waveform that results is difficult to interpret on an oscilloscope, especially if the signal frequencies are close to each other. The waveform in Figure 10-2 represents the output from a non-linear circuit that combines the input signals of 300 Hz and 500 Hz. The output waveform contains not only components of the 300 Hz and 500 Hz signals, but also a signal component at 800 Hz, and another signal component at 200 Hz.

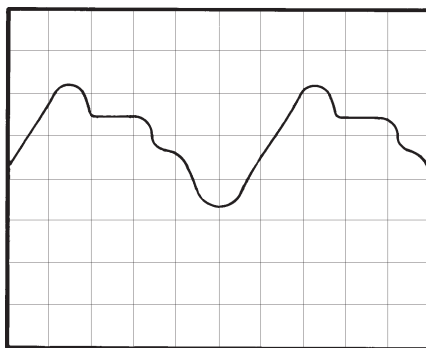


Figure 10-2

Signals in a complex waveform.

The 800 Hz and 200 Hz components are the sum and difference frequencies of the original 300 Hz and 500 Hz signals applied to the non-linear circuit. These sum and difference frequencies are the intermod products of the applied input signals.

Each cyclic variation in a complex signal waveform is a component of frequency within the waveform.

As with distorted waveforms containing harmonics, waveforms with intermod products contain the *components* of frequency in their waveform displayed on a oscilloscope. A more detailed, and accurate indication of these intermod signals can be obtained using a frequency-domain spectrum analyzer, just as for the analysis of harmonic signals. Look at these same signals on the spectrum analyzer display in Figure 10-3. The frequency domain display clearly shows the respective frequencies and power levels of the signals contained in the output waveform, or time domain, display. Just as with the harmonics, each of the intermod products are distinct sine waves.

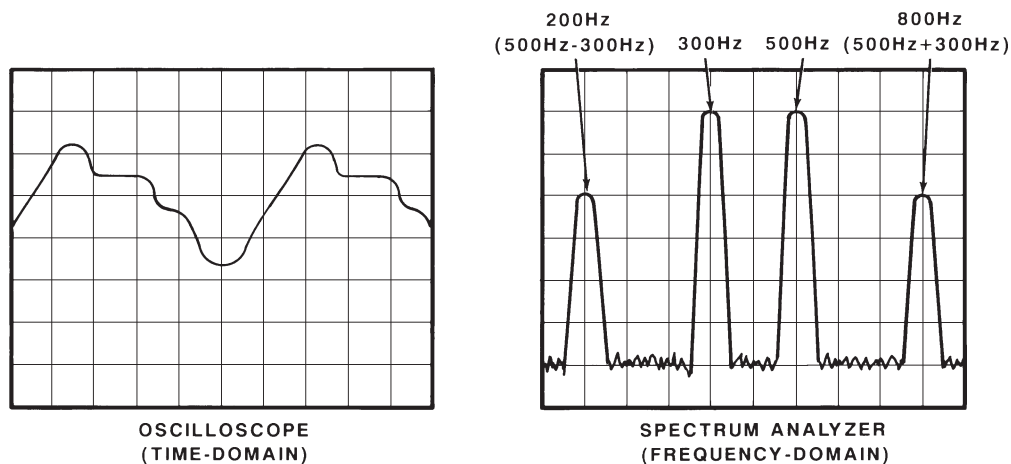


Figure 10-3

Frequency display using a spectrum analyzer.

FREQUENCY AND POWER DISTRIBUTION OF INTERMOD SIGNALS

Intermod Frequencies

Mathematically, the frequencies of intermod products are easy to calculate. They are simply the sum and difference of the signal frequencies that are applied to a non-linear network or circuit. For example, assume that two closely-spaced higher frequencies of 610 kHz and 1055 kHz are applied to the input of a non-linear circuit. The waveform at the output of the circuit will include the original 610 kHz and 1055 kHz signals, and also the sum and difference intermod products at 1665 kHz and 455 kHz. As an example, intermod circuits of this type, with these same applied frequencies, are used in virtually all AM broadcast radio receivers.

Intermod frequencies are the sum and difference of the signal frequencies processed through a non-linear network.

Now consider another example of two applied frequencies spaced much further apart. We'll select 610 kHz, and a much lower frequency of 2 kHz, applied to the input of a non-linear circuit. The waveform at the output of this circuit will include the original applied frequencies, along with the sum and difference intermod products at 612 kHz and 608 kHz. These two intermod signals are at frequencies only slightly above and below the 610 kHz signal. As an example, intermod circuits with these same applied frequencies are used in virtually all AM broadcast radio transmitters.

Intermod Distortion Factor

All signals that pass through non-linear circuits become distorted. Again, with single-frequency signals, the effects of distortion include the creation of harmonics. The power level remaining at the fundamental frequency, vs. the power distributed within the harmonic signals, is the measure of a circuit's non-linearity, its harmonic distortion factor. Recall that circuits with

harmonic distortion factors of less than 1% are considered good, and those with factors of less than 0.1% are considered near perfect. Recall also that the word “*distortion*” as applied identifies *unwanted* signals. When frequencies are wanted, they are described with words and names other than *distortion*; for example, “harmonic multiplier.”

Just as with harmonics, the effects of intermodulation result in the distribution of signal power at frequencies other than those applied to a circuit’s input. *And*, the power remaining (in the output waveform) at the input signal frequencies, vs. the power distributed within the intermod signals, is defined as a circuit’s intermodulation distortion (IMD) factor. Circuits designed for linear operation with IMD factors of less than 1% are considered good; those with IMD factors of less than 0.1 % are considered near perfect.

Intermodulation distortion (IMD) is a measure of the proportion of power distributed at intermod frequencies in the output waveform of a circuit.

The extent of intermod distortion depends upon the way in which a circuit allows signals to affect each other. With a maximum of interaction, the power distributed within the intermod frequencies can be as high as half that of the applied signals. For applications that take *advantage* of creating intermod frequencies, higher IMD factors are desirable. As with harmonics, there are some circuit functions where intermod frequencies are desirable, and there are other circuit functions where they are not wanted. The word “*distortion*” is usually used *only* when intermod is undesirable.

The measure of intermod distortion can be based upon the ideal linear circuit that produces zero intermod products. The signal power at the intermod frequencies can be calculated as the power ratio of all of the intermod signals (P_{IM}) vs. *one* of the two applied signals (P_{sig}), with both applied signals at the same power level.

$$\% \text{ IMD} = 3 \left(\frac{P_{IM}}{P_{SIG} + P_{IM}} \right) \times 100$$

The measurement of intermod power can best be made with a spectrum analyzer. The typical measurement set-up, and resulting spectrum analyzer display are shown in Figure 10-4. Figure 10-4B shows the applied signals directly connected to the spectrum analyzer; while Figure 10-4C shows the signals after they pass through the amplifier. The amplifier produces two intermod signals within the circuit's bandpass response. The output power levels of the intermod signals are 10 milliwatts each (for a total of 20 milliwatts of intermod power), while the output power level of each of the applied signals is 1 watt. The percentage of IMD can be calculated as follows:

$$\% \text{ IMD} = 3 \left(\frac{P_{\text{IM}}}{P_{\text{SIG}} + P_{\text{IM}}} \right) \times 100$$

$$\% \text{ IMD} = 3 \left(\frac{0.02}{1 + 0.02} \right) \times 100 = 5.9\%$$

In practice, the intermod signals are often *individually* specified, in dB, in reference to *one* of the applied signals. With the output power of the applied signal at 1 watt, and the power level of a *single* intermod signal at 10 milliwatts, the ratio of the two powers are:

$$\text{dB} = 10 \log \frac{P_{\text{IM}}}{P_{\text{SIG}}}$$

$$\text{dB} = 10 \log \frac{0.01}{1} = -20 \text{ dB}$$

The single intermod signal is 20 dB below the level of one of the applied signals, at the output of the circuit under test. This ratio is shown in Figure 10-4C.

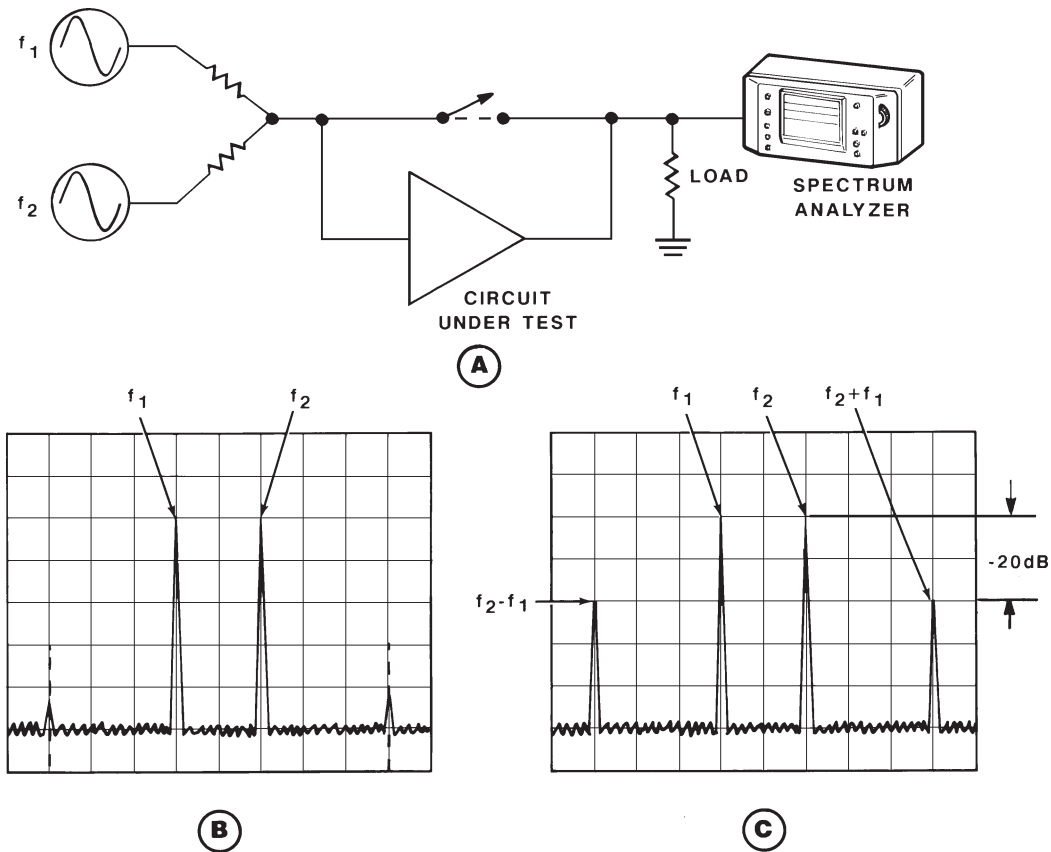


Figure 10-4
Measurement of intermodulation distortion.

Bandwidth

The preliminary description of intermod products was purposely simplified in order to provide you with a basic understanding of their frequency and power distribution, and their distinctions from harmonics. An accurate accounting of intermod signals is not so simple, however, as they seldom exist exclusive of harmonics. After all, the same non-linear circuits that create harmonics also create intermod products. While there are some exceptions, both harmonics and intermod frequencies occur simultaneously, as multiple signals are processed through non-linear circuits. The bandwidth, and true frequency distribution of these signals depends upon both harmonics and intermod products.

Recall that bandwidth is the term used to identify the range of frequencies occupied by a signal and its harmonics. With more than one signal simultaneously processed through a circuit, the bandwidth of the resulting output waveform includes all harmonic and intermod frequencies that are of a significant power level. Earlier, we specified a reference of 40 dB below the fundamental as our level of significance. (This dB figure varies depending upon the communication service or equipment to which it is applied, but we will use 40 dB as our “ballpark” figure.)

The bandwidth resulting from multiple signals processed through a circuit includes all significant harmonic and intermod frequencies.

Two signals of 1100 Hz and 1700 Hz are applied to a non-linear circuit. The signal waveform at the output of the circuit contains these two signals, plus their harmonics and intermod products. The spectrum analyzer display of Figure 10-5 shows the complete bandwidth of the original, harmonic, and intermod signals found at power levels above the -40 dB level of significance.

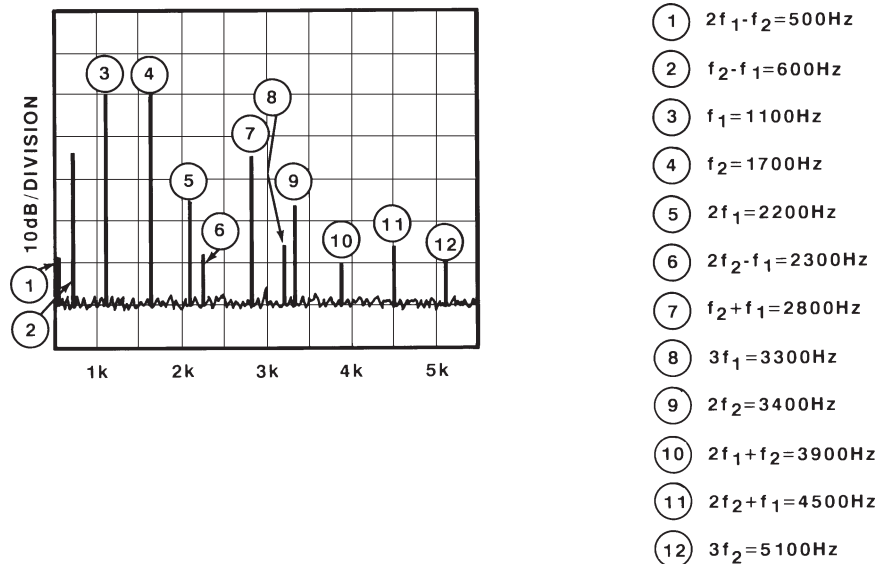


Figure 10-5

Distribution of intermodulation frequencies.

Using Fourier analysis, the frequencies and power levels of these signals can be calculated. However, the calculations are very theoretical, and often neglect the variables common to actual circuits. The words, “second-order,” “third-order,” etc, used in these descriptions, apply to frequency terms in the mathematical formula (the Taylor Series) used to identify them. You need not be skilled in higher math to extract and interpret the power levels of these frequencies, however, as they are usually *measured* quantities indicated by a spectrum analyzer. Each of the *frequencies* is easily calculated by adding and subtracting the intermod and harmonic values as follows:

- A. The two input signals, f_1 and f_2 , are at 1100 Hz and 1700 Hz respectively. These are the *first-order* terms.
- B. The intermod sum and difference frequencies of the two input signals, $f_2 + f_1$ and $f_2 - f_1$, are at 2800 Hz and 600 Hz respectively. These are the *second-order intermod* terms.
- C. The second harmonics of the two input signals, $2f_1$ and $2f_2$, are at 2200 Hz and 3400 Hz respectively. These are the *second order harmonic* terms.
- D. The intermod sum and difference frequencies of the *second harmonic* of f_1 are interacting with f_2 : $2f_1 - f_2$, which are 3900 and 500 Hz respectively. These are one pair of the *third-order intermod* terms.
- E. The intermod sum and difference frequencies of the *second harmonic* of f_2 are interacting with f_1 : $2f_2 - f_1$, which are 4500 and 2300 Hz respectively. These are the *other* pair of the *third-order intermod* terms.
- F. The third harmonics of the two input signals, $3f_1$ and $3f_2$, are at 3300 Hz and 5100 Hz respectively. These are the *third-order harmonic* terms.

As you may conclude, the process of identifying harmonic and intermod frequencies is much more complicated than identifying only harmonics. However, all frequencies (including the higher-order products) can be identified by the same *multiply-add-subtract* approach used in steps A through F.

The range of frequencies, through the third-order terms shown on the display of Figure 10-5, extends from 500 Hz through 5100 Hz. The frequencies characterize the signal waveform at the output of the amplifier of our example when two signals are applied to its input. The bandwidth of the amplifier’s output waveform is 4600 Hz ($5100 \text{ Hz} - 500 \text{ Hz} = 4600 \text{ Hz}$). A circuit with 4600 Hz bandpass, and a passband from 500 Hz to 5100 Hz, retains all of these intermod and harmonic signals within the waveform.

The power level of each of the harmonic and intermod signals is best obtained directly from the spectrum analyzer display. When you account for power, remember that *each* of the signals displayed by the spectrum analyzer has its own power level, and that all of the intermod signals are at *paired* (sum and difference) frequencies. To account for *all* of the fundamental, intermod, and harmonic power (total power), the power level of each of the individual signals must be summed.

The popular way of expressing the power levels of signal frequencies within the bandwidth of a complex waveform, is to reference them to the fundamental frequencies (first-order terms) of the applied signals. Looking back at Figure 10-5, you should observe that the second-order terms are between 10 and 20 dB below the level of the fundamental frequencies; and that the third-order terms are between 20 and 40 dB below the level of the fundamentals.

SUMMARY

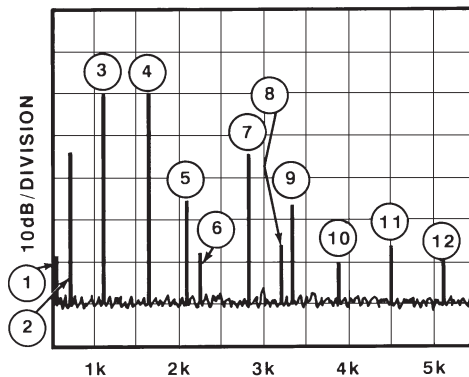
The combining of two or more signals in non-linear circuits results in effects much different than with linear circuits. In non-linear circuits, the signals interact with each other to create other signals in addition to those applied to the circuits. These other signals are called “Intermodulation products” (“intermod” for short). The intermod products occur at frequencies that are the sum and difference of the applied signals and their harmonics.

SELF-TEST REVIEW

1. When two signals are applied to a non-linear circuit, the *output waveform* will contain:
 - A. both signals and their harmonics.
 - B. both signals and intermod frequencies in addition to their harmonics.
 - C. both signals only.
 - D. harmonics and additional intermod frequencies only.
2. The *frequencies* that are created as a result of two signals interacting with each other as they pass through a non-linear circuit are called:
 - A. mixing products.
 - B. harmonic products.
 - C. intermodulation products.
 - D. non-linear distortion products.

3. A detailed and accurate indication of intermod signals can be obtained by using a/an:
- A. oscilloscope.
 - B. multimeter.
 - C. spectrum analyzer.
 - D. frequency counter.
4. Two signals are combined in a non-linear circuit. Within the same bandwidth, there will be _____ harmonics than intermod products.
more/fewer
5. A complex waveform includes harmonics and intermod frequencies. Through what circuit can this waveform be processed to retain *only* its lower-frequency component?
- A. Amplifier.
 - B. Rectifier.
 - C. High-pass filter.
 - D. Low-pass filter.
6. Two signals, at 900 and 1100 Hz, are processed through a non-linear amplifier. What are the first two intermod frequencies?
- 1st _____ Hz
- 2nd _____ Hz
7. Two signals, at 200 and 1100 Hz, are processed through a non-linear amplifier. What are the first two intermod frequencies?
- 1st _____ Hz
- 2nd _____ Hz
8. Two signals, at 900 and 1100 Hz, are processed through a non linear amplifier. What are the four *third-order* intermod frequencies?
- A. _____ Hz
 - B. _____ Hz
 - C. _____ Hz
 - D. _____ Hz

9. Two signals, at 900 and 1100 Hz, are processed through a non-linear amplifier. If the passband of the amplifier is from 100 to 3000 Hz, How many harmonic and intermod frequencies will exist within the amplifier's output waveform? _____
10. Two signals are applied to a non-linear circuit. The power levels of these signals are the same 0.1 watt (100 milliwatts) at the circuit's output. The power level of the intermod signals at the circuits output total 5 milliwatts. What is the *percentage* of intermodulation distortion? _____ %
11. Two signals are applied to a non-linear circuit. The power levels of these signals are the same 0.1 watt (100 milliwatts) at the circuit's output. The power level of *one* of the intermod signals at the circuits output is 2 milliwatts. How many *dB* below the output signal is the intermod signal? _____ dB
12. In Figure 10-6, how many dB below the applied output signal is the power level of the largest third-order intermod signal? _____ dB



- ① $2f_1 - f_2$
- ② $f_2 - f_1$
- ③ f_1
- ④ f_2
- ⑤ $2f_1$
- ⑥ $2f_2 - f_1$
- ⑦ $f_2 + f_1$
- ⑧ $3f_1$
- ⑨ $2f_2$
- ⑩ $2f_1 + f_2$
- ⑪ $2f_2 + f_1$
- ⑫ $3f_2$

Figure 10-6

Figure for Question 12.

CHAPTER 11 — Applications of Intermodulation Distortion

INTRODUCTION

Earlier, we introduced the fundamental concepts of intermodulation distortion, and described how intermod frequencies are created as two or more signals pass through non-linear circuits. Then, we showed how harmonic and intermod products can complicate the number of frequencies that exist within the bandwidth of a complex signal's waveform. In this section, we will describe how specific harmonic and intermod frequencies can be used to advantage in signal processing functions within communications equipment.



OBJECTIVES

When you complete this chapter, you will be able to:

- Specify the effects of bandpass and frequency response upon the signal waveforms and intermodulation frequencies.
- Determine the effects of phase upon the generation and suppression of harmonic and intermod signals.
- Determine the effects of filtering upon the generation, suppression, and/or selection of harmonic and intermod signals.
- Apply the concepts of intermod products to the modulation of signals in communication systems.
- Apply the concepts of intermod products to the frequency conversion of signals in communication systems.

SUPPRESSING HARMONIC AND INTERMOD SIGNALS

From your study of Chapters 9 and 10, you should have concluded that there are numerous frequencies created within non-linear circuits. For many signal processing functions, some of these frequencies must be suppressed, while others are enhanced. There are a number of ways to reduce harmonic and intermod signals within a waveform. The more popular techniques include the following:

- A. One choice is to make the circuits through which signals pass linear. Neither harmonics nor intermod products are created in linear circuits.
- B. Another choice is to use critical filtering. Reducing the bandpass of a circuit's output will selectively reduce the range of frequencies in the output signal's waveform.
- C. A third choice is to phase-cancel signals. A signal can be suppressed by combining it with an identical waveform that is out-of-phase with the original signal.

There are many applications in communications technology that either purposely create and/or suppress harmonic and intermod frequencies. Keep in mind that harmonics *alone* are created when *only* one signal is processed through a non-linear circuit; and that *both* harmonic and intermod frequencies are created when more than one signal is processed through a non-linear circuit. The applications you will study in this section depend upon the creation of selected intermod frequencies. Consequently, the technique of using linear circuitry, choice A, is not a valid option. Choices B and C, however, are used extensively in almost all non-linear circuit applications.

MODULATION

One of the most popular applications of non-linear circuits, and using frequency selection and suppression, is modulation. *Modulation* is the process of applying an information signal to a higher-frequency signal to take advantage of the higher-frequency signal's special properties. When so used, the higher-frequency signal is called a "carrier" (a carrier of information). Information signals are lower in frequency, and include the waveforms of voice, music, video, and/or data. Carrier signals are higher in frequency, and have the advantages of increased bandwidth and transmission capabilities.

Modulation is the process of applying an information signal to a higher-frequency carrier signal to take advantage of the carrier signal's bandwidth and transmission properties.

A good example of efficient *carriers* are microwave signals. Typical satellite microwave signals can simultaneously *carry* thousands of voice and data *information* signals, or up to 30 television channels, due to their extended bandwidth capacity. And these microwave satellites efficiently beam their signals from 22,000+ miles above the earth, using only about 10 watts of power. Compare this 10 watts to the 1,000 watts it takes to make breakfast toast, and the efficiency of such carrier signals becomes quite obvious. NOTE: The newer direct-broadcast satellites transmit on the order of 100 to 300 watts, but allow you to use a much smaller dish to receive them.

There are many modulation forms. Probably those most familiar to you are AM (amplitude modulation) and FM (frequency modulation), because of your everyday contact with broadcast radio services that use these modulation forms. There are many more modulation forms applied within telecommunication systems (including the telephone) that you use regularly. Most of these forms are not popularly identified, so it is not likely that you may know them. As you continue your studies of communications technology, however, you are sure to become familiar with ten, or, more, different modulation forms. We will not attempt the complicated task of describing all modulation forms in this text, but you will have *the foundations* of their operation if you understand the principles of intermodulation. In describing the most basic modulation forms in the following pages of this text, we will emphasize their operation as an application of the intermodulation process.

Amplitude Modulation

The amplitude modulation (AM) process is the simplest way to apply an information signal to a carrier signal. It is a process by which the amplitude of an information signal controls the amplitude of a carrier signal. The result is also the intermodulation distortion of the carrier signal by the information signal. The carrier and information signals are applied to a non-linear circuit, in this case called a “modulator” circuit. The modulator produces a carrier signal that has been affected by the information signal. This creates intermod frequencies, and the *selection* of the second-order intermod products to accompany the carrier in the waveform of the combined signals.

Amplitude modulation is a process by which the amplitude of an information signal controls the amplitude of a carrier signal.

Figure 11-1 shows the typical application of the modulation process as applied to common AM broadcast radio. The voice or music *information* signals occupy frequencies up to 5 kHz, while the radio frequency (RF) carrier signal is in the range of 1000 kHz. In the modulation process, the first pair

of sum and difference intermod frequencies occur on either *side* of the carrier, and only these three frequencies pass through the output filter of the circuit. The lower-frequency component, the information signal, is filtered out. Also, the harmonics and higher-order intermod products of the two frequencies are filtered out of the signal waveform. The *desired* signals are selectively filtered with all other frequencies suppressed. This example of the modulation process is typical of those used in virtually all AM communications systems.

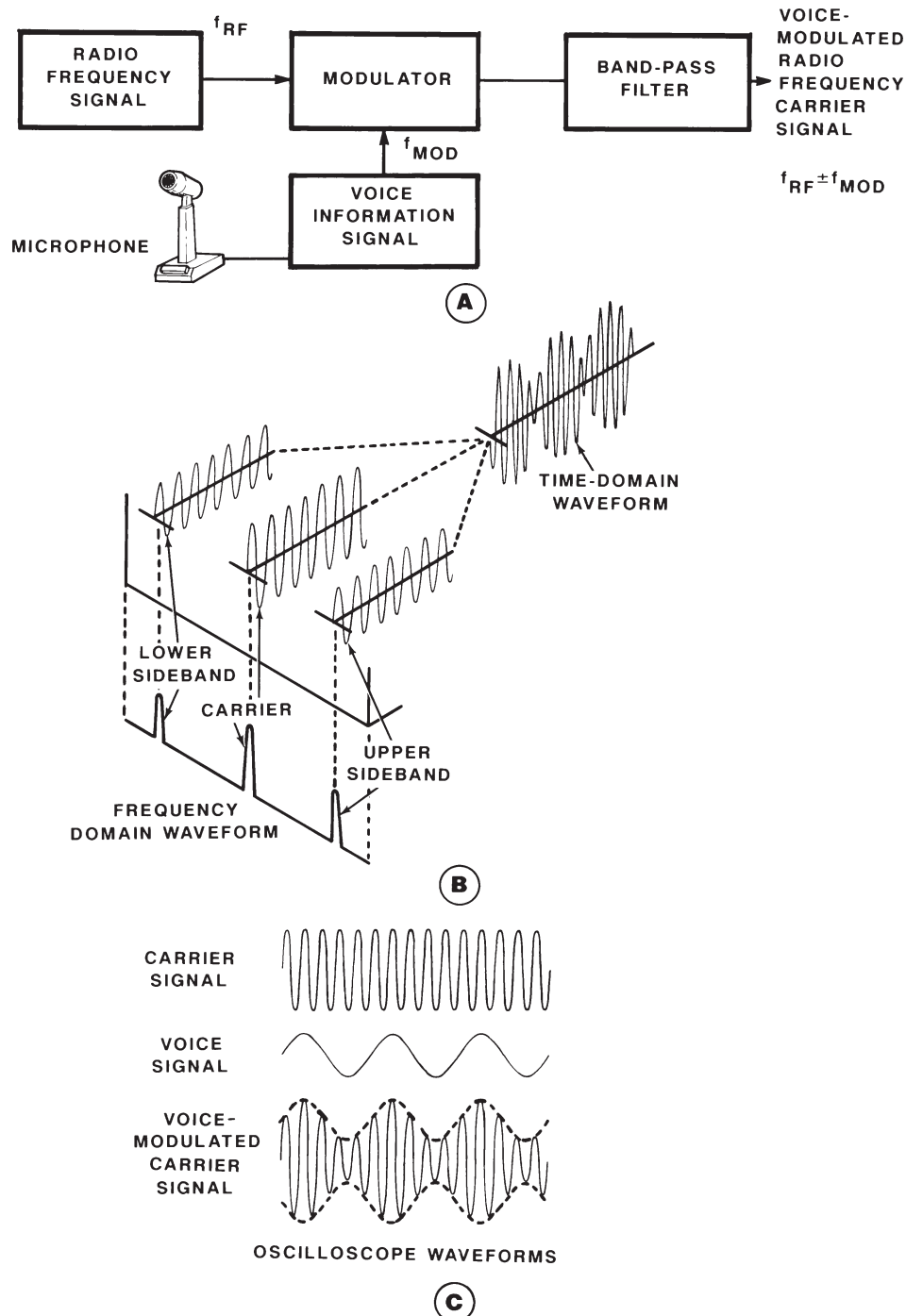


Figure 11-1
Amplitude modulation.

The intermod frequencies, 5 kHz above and below the carrier, as shown in Figure 11-1B, are called the “upper sideband (USB) and lower sideband (LSB) of the carrier signal.” The passband of the circuit’s output filter includes the sidebands, and extends from 995 kHz (1000 kHz – 5 kHz) to 1005 kHz (1000 kHz + 5 kHz). The time-domain oscilloscope waveform of Figure 11-1C shows the typical display obtained from the AM signal. Compare this waveform with that of Figure 9-2 (on page 9-5) to see the clear distinctions between signals which are combined in linear and non-linear circuits.

“AM” identifies a *process* of modulation. It is one of many modulation processes used in telecommunications systems. This modulation process includes a carrier frequency and its two adjacent (upper and lower) sideband signals in its output waveform. By definition, AM is more accurately identified as double-sideband amplitude modulation (abbreviated DSB-AM).

(Students often confuse AM with that range of frequencies used for AM broadcast radio. This is not the case. AM is used even for television broadcast and aircraft communication, at frequencies far removed from the AM radio broadcast band!)

AM is the abbreviation for “amplitude modulation,” which identifies a process of modulation, whereby a carrier and two adjacent sideband frequencies are included in its output waveform.

Sideband Power Distribution

The power distributed within the sideband frequencies of the AM waveform are identical to the power distribution you studied earlier to determine intermodulation distortion (IMD). However, the *approach* used to calculate the power of *these* sideband frequencies differs from that used to calculate IMD. The power in the sideband (intermod) frequencies of AM waveforms are traditionally calculated using a modulation index factor. This factor relates the varying level of the *information* signal, to the fixed level of the carrier signal. (Recall that IMD calculations assume the levels of both applied signals are identical.) The modulation index for the AM waveform (M_{AM}) can be calculated as:

$$M_{AM} = \frac{V_{mod}}{V_c} \text{ and } \%M_{AM} = \left(\frac{V_{mod}}{V_c} \right) 100$$

V_{mod} is the voltage level of the applied modulation (information) signal, and V_c is the voltage level of the carrier signal. With both signals at the same level (as for calculating IMD) the modulation index will be 1.0, or 100%. With the modulation signal voltage at half that of the carrier signal voltage, the modulation index will be 0.5, or 50%.

Once the modulation index has been determined, the power in the sideband (intermod) frequencies (PSB) can be calculated using the index factor that references the power level of the modulation signal to that of the carrier signal.

$$P_{\text{SB}} = P_c \left(\frac{M_{\text{AM}}^2}{2} \right)$$

Using this formula, and assuming a 1 watt carrier signal, the sideband (intermod) power can be calculated for the modulation index factors of 1.0 and 0.5, as follows:

for a M_{AM} of 1.0:

$$P_{\text{SB}} = 1 \left(\frac{1^2}{2} \right) = 0.5 \text{ W}$$

for a M_{AM} of 0.5:

$$P_{\text{SB}} = 1 \left(\frac{0.5^2}{2} \right) = 0.125 \text{ W}$$

Observe that the total sideband power (power level of *both* sideband signals) is *half* the power level of the carrier signal when both modulation and carrier signals are at the same level. With further mathematical analysis, it can also be shown that the total sideband power is half of the power of the modulating signal. In practice, however, modulation index and carrier power, not modulating signal power, are the references used to determine sideband power.

Sideband Modulation

There are two popular forms of sideband modulation: double sideband suppressed-carrier (DSB-SC), and single-sideband (SSB). Both of these modulation forms are similar to AM, but with the carrier frequency suppressed (removed). These popular modulation forms use selective *filtering* and *phase cancellation* to obtain the desired frequencies within their output waveforms. Each form has distinct advantages in telecommunication systems.

One way to understand the sideband modulation forms is to compare them with the AM (DSB-AM) spectrum analyzer display shown in Figure 11-2A. You should observe that the display of Figure 11-2B is identical to that of Figure 11-2A, with only the carrier signal missing. Figure 11-2B is the DSB-SC display. Figure 11-2C is the same as Figure 11-2A, except the carrier and one of the sideband signals are missing. Figure 11-2C is the SSB display.

DSB-SC and SSB are modulation forms that are similar to AM, but with the carrier removed.

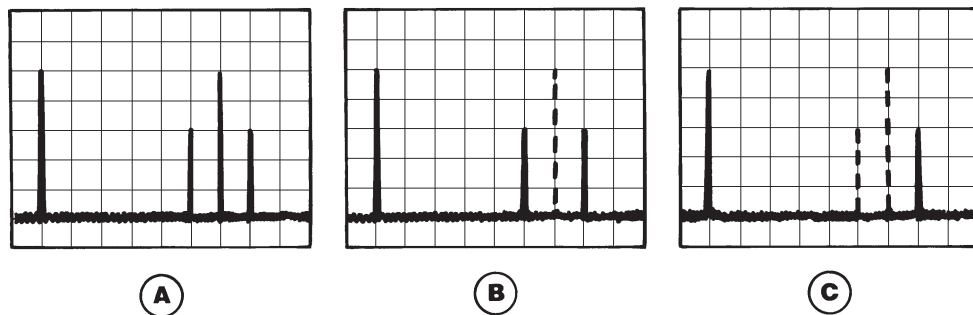


Figure 11-2

AM, DSB-SC, and SSB.

One of the advantages of DSB-SC and SSB over conventional AM is power efficiency. The sideband frequencies contain all of the modulation information, so it is not entirely necessary to transmit a signal with the carrier included. With 100% modulation (modulation index = 1.0) of a 1-watt carrier, for example, there will be 0.5 watts of power within the two sideband frequencies, and a total power of 1.5 watts consumed. The *same* signal information can be communicated with DSB-SC using only 0.5 watts (the power of two sideband signals), or with SSB using only 0.25 watts (the power of one sideband signal).

The additional advantage of SSB over both of the other modulation forms is bandwidth. With SSB, the same modulation information occupies only half the range of frequencies occupied by AM and DSB SC. If the modulation *information* is applied over a range of frequencies from 300 Hz through 3000 Hz, for example, those *same* frequencies will also exist within the sideband frequencies of the carrier. With AM and DSB-SC, there will be a total of 6000 Hz bandwidth occupied by the modulated signal. With SSB, the same information requires an bandwidth of only 2700 Hz (300 Hz to 3000 Hz).

Double-Sideband Suppressed-Carrier

With DSB-SC, the modulating information signal is filtered from the signal's output waveform, just as with AM. An *advantage* of suppressed-carrier modulation is that it eliminates higher-order harmonic and intermod products. Therefore, the suppressed-carrier circuits don't require the extensive filtering that is necessary with AM.

An advantage of suppressed-carrier modulation is that it eliminates higher-order harmonic and intermod products without extensive filtering.

Recall that harmonic frequencies are multiples of the carrier frequency; and that higher-order intermod frequencies result from the modulation information signal interacting with the *harmonics* of the carrier. If the carrier is suppressed, so too will its harmonics be suppressed. And, without interaction of the harmonics, the intermod frequencies are also suppressed. So, one significant advantage of carrier suppression is elimination of the extensive filtering of frequencies beyond the second-order intermod products.

The carrier signal is suppressed using phase cancellation. This is accomplished with a balanced modulator that uses diodes as the non-linear elements, as shown in Figure 11-3A. In this circuit, the diodes must be carefully matched to achieve a good carrier balance, or null.

Figure 11-3B shows the circuit operation when there is no audio input and the carrier input has a polarity as shown. Diodes D1 and D2 are forward-biased, and equal, but opposite, currents flow in both sides of T1. Therefore, there is no output. When the carrier input polarity reverses, diodes D3 and D4 conduct, as shown in Figure 11-3C. Here again, equal but opposite, currents flow in T1, and there is no output.

Figure 11-3D shows the circuit action when the audio signal is applied. To simplify the drawing, the carrier input has been removed and the diodes replaced by switches. When diodes D1 and D2 conduct, due to the carrier input, the audio signal flows through T1 in the direction shown. When the carrier signal changes polarity, as shown in Figure 11-3E, diodes D3 and D4 conduct. This causes the signal through T1 to change polarity. Therefore, the audio polarity at T1 is actually being switched back and forth by the carrier input.

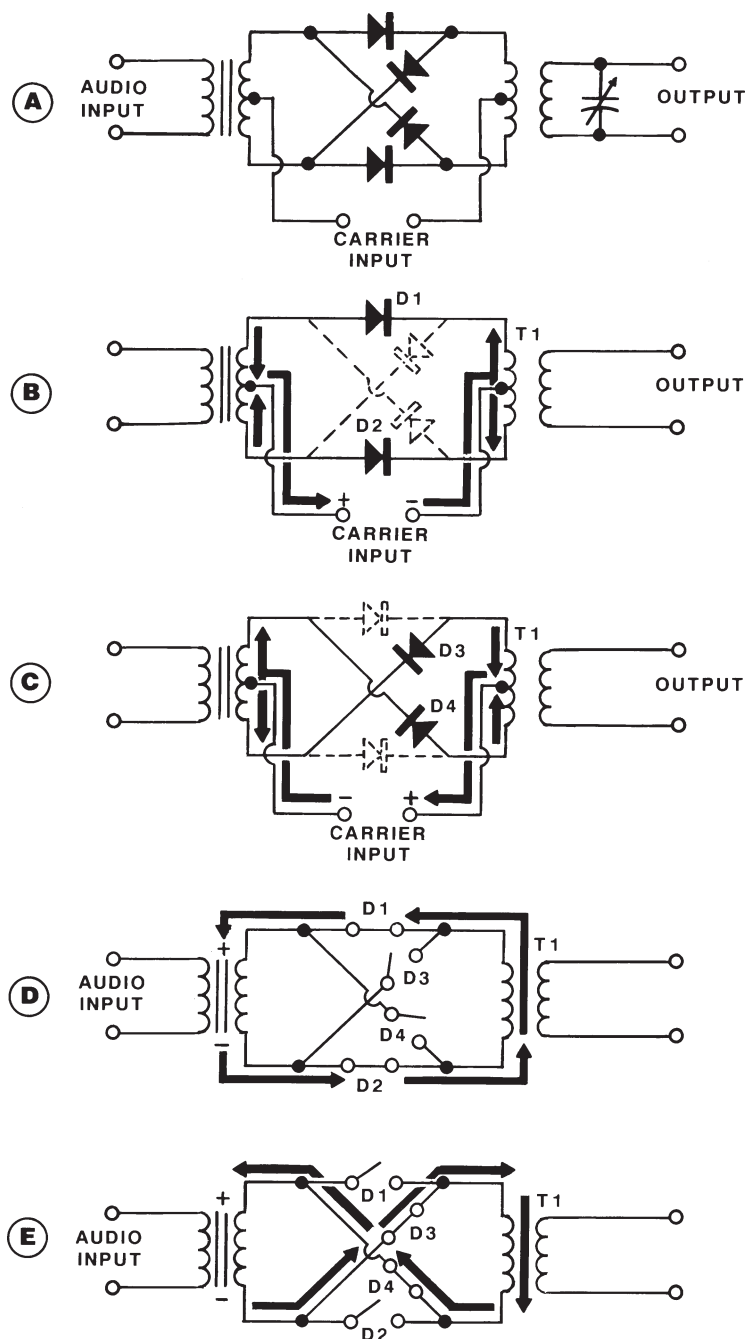


Figure 11-3
Balanced modulator.

To clarify matters, the audio signal does not actually forward bias the diodes. Instead, the RF input switches the diodes and the audio signal sees the forward biased diodes as a low-impedance path. This is caused by the varactor action of the diodes. A forward-biased diode has a high capacitance and, therefore, a low capacitive reactance to the audio signal. On the other hand, the reverse-biased diodes offer a high reactance to the audio signal.

Balanced modulators can also be configured using differential amplifier circuits, which are available as IC devices. Amplitude modulation cannot be obtained by using a balanced modulator; balanced modulators cancel the carrier signal. AM requires the use of an unbalanced modulator. Conversely, suppressed carrier modulation can be obtained by using a balanced modulator to cancel (suppress) the carrier signal.

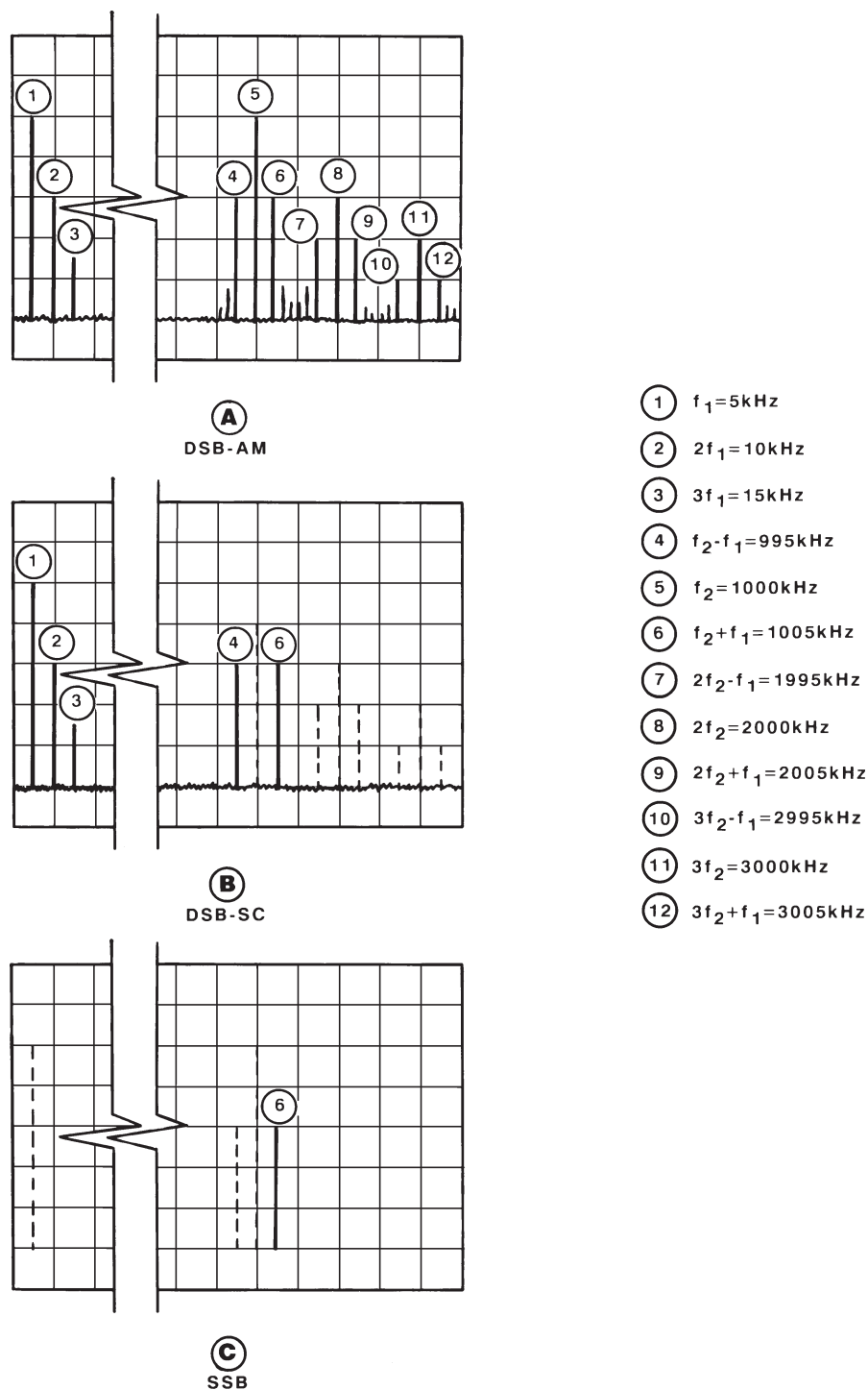
The intermod products of two signals, the upper and lower sideband signals, are initially out-of-phase with each other. That is, the sum frequency (upper sideband signal) is 180° out-of-phase with the difference frequency (lower sideband signal). (The sideband signals are + and – in frequency and phase.) Even though the sideband signals are produced within the common non-linear network that phase-cancels the carrier, they remain distinct and are not phase-canceled. The sideband intermod products are a result of the interaction of the applied carrier signal with the modulating information signal, not after carrier cancellation. And being out-of-phase, they are not affected by the phase shifting of the carrier.

Figure 11-4A shows many frequencies indicated on the spectrum analyzer display. These are the frequencies created by the amplitude modulation of a 1000 kHz carrier signal with a 5 kHz information signal. For a passband from 4 kHz through 3000 kHz, for example, no less than a dozen frequencies exist within the output waveform of the non-linear circuit. The modulating frequency and its harmonics can be rejected with a simple high-pass filter at the circuit's output. The higher frequencies (above 1005 kHz) can likewise be rejected with a low-pass filter. With DSB-SC modulation, the higher-frequency signals are eliminated, as shown in Figure 11-4B, without filtering.

Single Sideband

Figure 11-4C shows the results of phase-canceling and *additionally* filtering one of the two sideband frequencies from the output waveform of a non-linear circuit. The result is the single-sideband (SSB) modulation form. As mentioned earlier, the advantages of SSB include *both* power efficiency *and* bandwidth efficiency. SSB is also a suppressed-carrier modulation form, so like DSB-SC, it too eliminates higher-order harmonic and intermod products. Single sideband is the *most efficient* form of modulating voice information on carrier signals. It is, therefore, one of the more common forms of modulation used in telecommunication systems.

The advantages of SSB include both power efficiency *and* bandwidth efficiency.

**Figure 11-4**

Frequency distribution of harmonic and intermod products
with AM, DSB-SC, and SSB.

For SSB modulation, the carrier signal is suppressed through phase cancellation within a non-linear circuit, as with DSB-SC. *One* of the sideband frequencies (upper or lower) is then processed through a high-Q *band-reject* filter; while the other sideband frequency is processed through a high-Q, *bandpass* filter. The single frequency at the output of the *bandpass* filter is that of the SSB signal.

Assume an example of modulation typical of those used in multiplex telephony. A modulating *voice* information signal occupies frequencies between 300 Hz and 3300 Hz. This signal is to be SSB modulated on a 20 kHz carrier. The signals are combined in a circuit arrangement similar to that shown in Figure 11-5. (You may recall that an expanded version of this circuit was introduced in Figure 8-3 of Chapter 8, which described an application of harmonics and filters.) With suppressed-carrier modulation, the filters *pass* the upper sideband and *reject* the lower sideband of the signal waveform they process. The output from the bandpass filter includes frequencies from 20.3 kHz to 23.3 kHz.

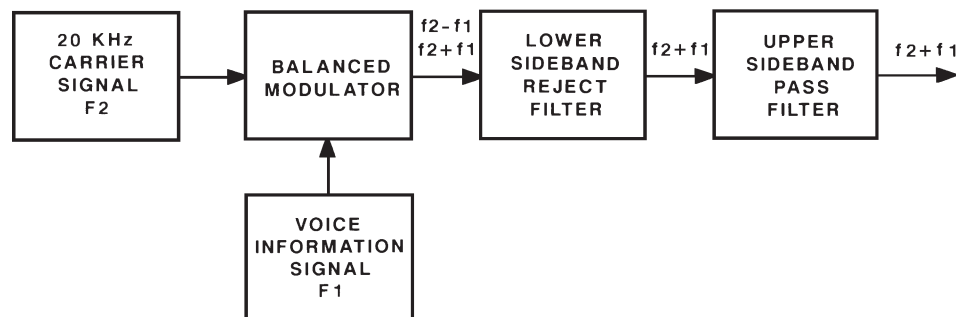


Figure 11-5

SSB balanced modulation.

If the power level of the modulating voice signal is equal to the input power level of the 20 kHz carrier (before suppression), a modulation index of 1.0 will exist, and the power in *both* sidebands will be half the carrier power ($M_{AM}^2/2$). The power within the upper sideband passing through the filter will be *half again* that of both sidebands, which is one-fourth the power level of the modulating voice signal. Put more simply, the power level of the SSB signal is always one fourth that of the modulating voice signal; that's why the levels of SSB signals are often called "talk power." Their power levels vary in direct proportion to that of the modulating voice signal applied to the SSB modulator.

With AM, DSB-SC, and SSB, the power level of each of the sideband signals is one-fourth the power level of the modulating information signal.

Yet another advantage of the SSB modulation form is the availability of either sideband for the multiplexing of voice signals. One voice signal can be modulated as an *upper* sideband signal, while a second voice signal can be modulated as a *lower* sideband signal; doubling the information capacity of each suppressed carrier frequency. You may be familiar with this technique as it is used for Citizen's Band and amateur radio, but it is also used in virtually all telecommunications systems. Often, SSB communications receiving equipment has front-panel switches that allow the operator to select and process either (lower or upper) of the sideband signals.

As mentioned earlier, there are many different modulation forms in addition to the three (DSB-AM, DSB-SC, and SSB), described in this text. However, the principles introduced here will apply to all forms of modulation. The modulation is a process of combining signals in a non-linear circuit which creates harmonic and intermod frequencies. The resulting frequencies can be selected and/or suppressed using filter and phasing techniques. Your understanding of the modulation processes described in this section can be effectively applied to your future studies of the other modulation forms.

MIXERS

Modulation is an important application of intermod distortion used in electronic communication. Non-linear mixing is another *equally important* application of intermod frequencies that is also used extensively in telecommunications equipment. Earlier, we made distinctions between combining signals in linear vs. non-linear circuits. We explained how combining two or more signals in non-linear circuits creates harmonic and intermod products. The SSB modulation form was described as a process that uses only *one* of the two intermod frequencies to carry signal information.

Like SSB modulation, the *mixing* process also uses only one of the intermod frequencies to carry signal information. There is a small, but significant difference, however, between mixing and modulation. The major distinction between the mixing and modulation processes is simply a matter of *purpose*. The purpose of modulation is to use intermod frequencies to carry information; while the purpose of mixing is to change the frequency of the information being carried. Non-linear mixing is primarily used for frequency conversion.

The purpose of modulation is to use intermod frequencies to carry information; while the purpose of mixing is to change the frequency of the information being carried. The primary purpose of non-linear mixing is frequency conversion.

The spectrum analyzer displays of Figure 11-6 compare the intermod frequencies resulting from the *modulation process*, to those of the *mixing process*. In the modulation process, the applied *information signal* is at a relatively low frequency, resulting in intermod products closely spaced above and below (in the “sidebands” of) the applied *carrier signal*. In the mixing process, the *two applied signals* are at frequencies closer to each other, and the resulting intermod products are spaced further apart, beyond the applied signal frequencies.

For example, let’s put some typical numbers on the frequencies applied to the *modulator* circuit of Figure 11-6A and displayed in Figure 11-6B, 3 kHz for the modulating information signal, and 60 kHz for the carrier signal. The resulting upper sideband output signal is at 63 kHz. By contrast, the two signals applied at the inputs of the *mixer* circuit of Figure 11-6C, 60 kHz and 48 kHz, are both higher in frequency. The resulting intermod output frequencies from the mixer circuit are at 108 kHz (the *sum frequency*) and 12 kHz (the *difference frequency*).

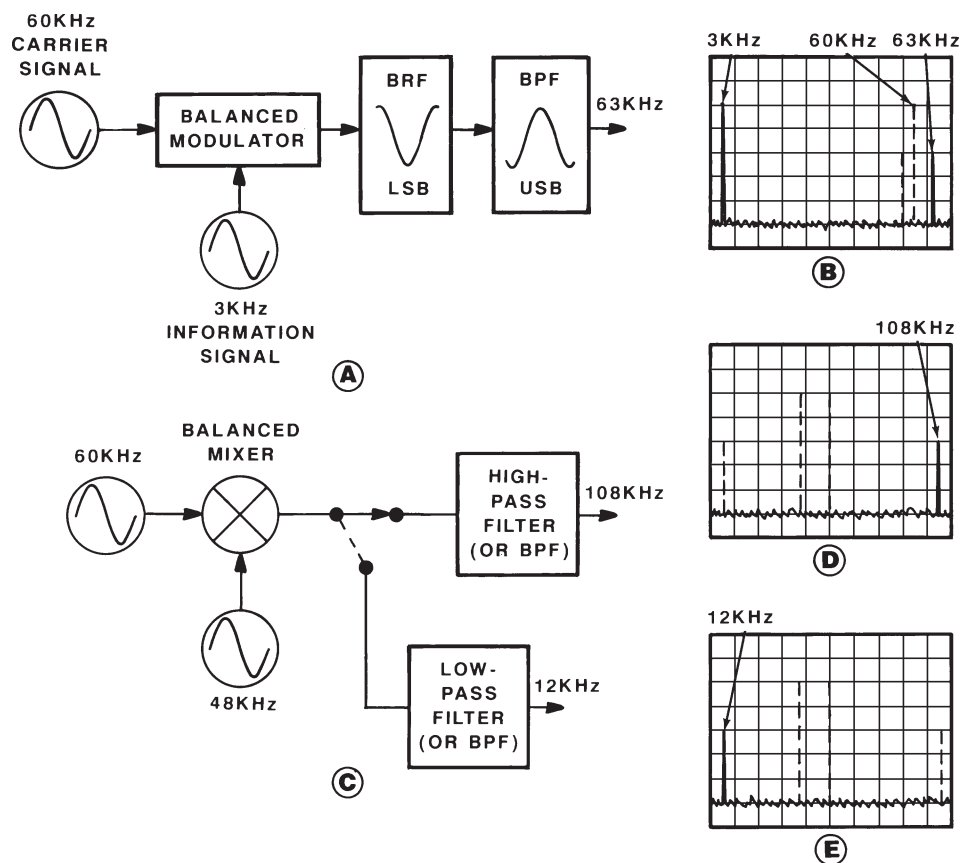


Figure 11-6

Balanced modulator and balanced mixer.

If the mixer output signal is passed through a simple *high-pass* filter, only the 108 kHz signal will exist. *One* of the input signals will have been frequency-converted, “up-converted”, to 108 kHz by the *other* input signal. Conversely, if the mixer output signal is passed through a simple *low-pass* filter, only the 12 kHz signal will exist. Again, *one* of the input signals will have been frequency converted, “down-converted”, to 12 kHz by the *other* input signal.

Frequency Conversion

In practice, frequency conversion is obtained by applying a *second* signal to the input signal within the mixer circuitry. That second signal is usually generated by an oscillator that is adjacent to the mixer circuit. Because the oscillator is in the same *locale* as the mixer, it is usually called a “local oscillator” (LO). Figure 11-7 shows the typical block representations and the mixer symbol of the frequency converter circuitry.

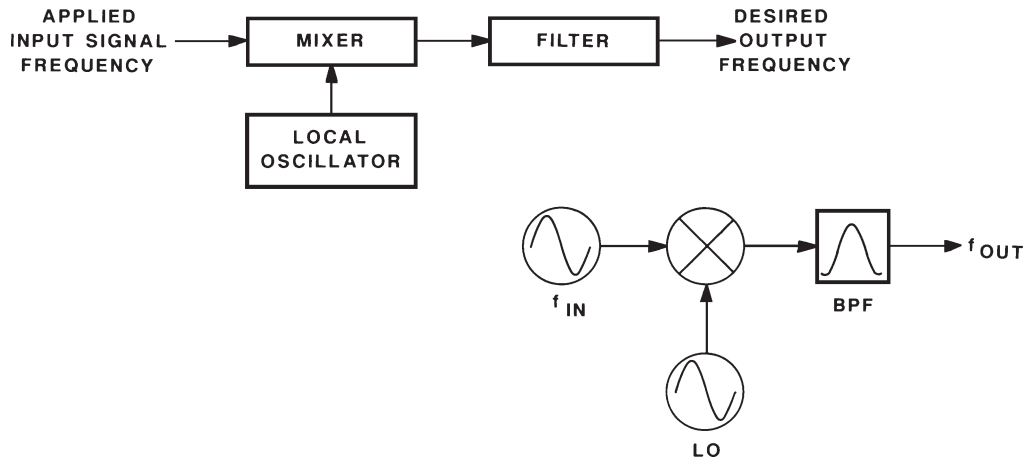


Figure 11-7

Frequency conversion.

Frequency conversion is obtained by combining a local oscillator signal and an applied input signal within a non-linear mixer circuit.

The frequency of the LO determines the frequency to which the applied input signal is to be converted. Conceptually, the LO can be set to *any* frequency, which in turn, allows the output of the frequency converter to be set to *any* frequency. Any modulation or other properties of the applied input signal will be carried through the mixing process, and remain a part of the mixer's output frequency. To assure that no additional frequencies, or distortion products are added to the applied input signal, the LO signal *must* be a "clean" sine wave, with *very* low harmonics within its waveform. The following are examples of frequency conversion, typical of those found in telecommunications circuits.

- A. A 108 kHz signal can be *up-converted* to exactly 552 kHz by mixing it with a 444 kHz LO signal. 552 kHz is the *sum* of 444 kHz and 108 kHz.
- B. The 108 kHz signal can also be *up-converted* to 552 kHz by mixing it with a 660 kHz LO signal. 552 kHz is the *difference* between 660 kHz and 108 kHz.
- C. A 3 kHz signal can be *up-converted* to exactly 64 kHz by mixing it with a 61 kHz LO signal. 64 kHz is the *sum* of 61 kHz and 3 kHz.
- D. The 3 kHz signal can also be *up-converted* to 64 kHz by mixing it with a 67 kHz LO signal. 64 kHz is the *difference* between 67 kHz and 3 kHz.
- E. A 312 kHz signal can be *down-converted* to exactly 71 kHz by mixing it with a 241 kHz LO signal. 71 kHz is the *difference* between 312 kHz and 241 kHz.
- F. The 312 kHz signal can also be *down-converted* to 71 kHz by mixing it with a 383 kHz LO signal. 71 kHz is the *difference* between 383 kHz and 312 kHz.

In studying examples C and D you may observe that the frequency conversion process is virtually identical to that of SSB modulation. The two intermod frequencies are like sideband signals, and high-Q filters will be required to pass *only* the sum (example C) or difference (example D) frequency to the frequency converter circuit's output. Keep in mind the distinctions, however: SSB modulation is a process of applying an information signal to a carrier; while mixing is the process of changing the frequency of any signal (in this case, the information signal) to some other frequency.

Further study of examples A vs. C (or B vs. D) should disclose that up-conversion of a low-frequency signal results in closely-spaced intermod frequencies. When frequencies are closely spaced, high-Q filters are required to select the desired mixer output signal.

Also, as you study and compare examples E and F to the other examples, you should note that down-conversion occurs *only* with difference frequencies, but two *different* frequencies will provide the same resulting output signal.

Historically, the process of non-linear mixing was called “heterodyning.” “Superheterodyne” is the word used to describe the mixing process when the LO frequency is *higher* than that of the applied input signal. Examples B, D, and F are examples of superheterodyne frequency conversion.

“Subheterodyne” is the word used to describe mixing the applied input signal with that of a LO at a frequency *lower* than the input signal. Examples A, C, and E are subheterodyne frequency conversions, while examples B and D are superheterodyne frequency conversions.

Balanced and Unbalanced Mixers

Like the unbalanced and balanced modulators used for AM and sideband modulation forms, unbalanced and balanced mixers are available to perform a number of specific frequency conversion tasks in communications systems.

An unbalanced mixer is the simplest of designs, but it produces all of the harmonic and intermod frequencies at its output. There are too many frequencies, and they are too closely spaced to practically select and reject with filtering.

Mixers are like modulators. The output waveforms of unbalanced mixers contain their input signals, and the harmonic and intermod frequencies of these signals. The output waveforms of balanced mixers include only the sum and difference intermod frequencies.

Unbalanced mixers are more practical when their input signals are moderately spaced in frequency. The frequency conversion with example F, on page 11-17, will work well with an unbalanced mixer. With input signals at 312 and 383 kHz, and a desired difference frequency of 71 kHz, the sum

frequency will be 695 kHz. The closest higher order intermod product frequency will be at 141 kHz ($2f_1 - f_2$), well beyond frequencies of concern for filtering.

In all cases, you need to be aware of the harmonic and intermod products created with unbalanced mixers. You must account for all signals, to determine filter requirements, and assure that unwanted frequencies will not interfere with other signals. The major feature of unbalanced mixers is their simplicity. Virtually any non-linear circuit that produces intermod products can function as an unbalanced mixer. Figure 11-8A shows such a simple circuit using only a diode, resistor, and output filter (LC resonant network).

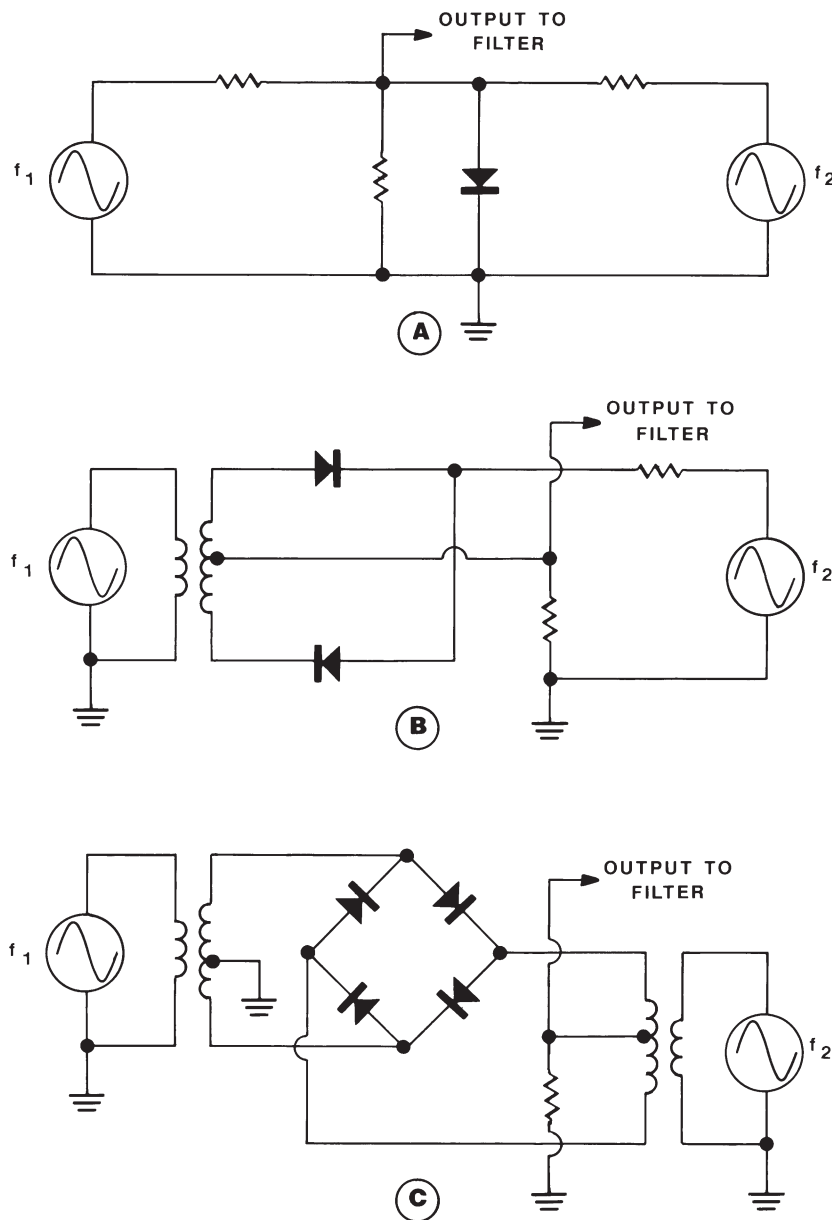


Figure 11-8

Unbalanced, single-balanced, and double-balanced mixers.

Unbalanced mixers are generally impractical for use with up-conversion of low-frequency signals. With the earlier frequency conversion example C, on page 11-17, the output of an unbalanced mixer will include the desired 64 kHz, but it will also include frequencies of 58 kHz, 61 kHz, 119 kHz, 122 kHz, and 125 kHz, among others. The frequencies adjacent to the desired 64 kHz signal will be difficult to filter, and the higher-order frequencies may interfere with other signals.

Balanced mixers are far more practical for use as frequency converters than unbalanced mixers, as they suppress one or both of the signals applied to their inputs. Like the balanced modulators, suppressing these signals will *also* suppress their harmonics and higher-order intermod products. *Only* the sum and difference frequencies of the input signals will remain within the output waveforms of balanced mixers.

Figures 11-8B and 11-8C show single- and double-balanced mixers, respectively. With single-balanced mixers, only *one* of the input signals is balanced (suppressed). With double-balanced mixers, however, *both* of the input signals are balanced. Balance (suppression) is obtained in these circuits by center-tapping the input transformer sections to which the mixer signals are applied. The opposing currents flowing through the common networks of the circuits result in cancellation of the input signals, while the diodes serve to make the circuits non-linear.

Single-balanced mixers are generally used for the up-conversion of low-frequency signals, such as voice signals. The harmonics of the low-frequency signals will not usually extend into the frequency range of the mixer's output frequency, so their suppression is not necessary. A single-balanced mixer is suitable for our earlier frequency conversion examples C and D, on page 11-17, as the 3 kHz signal is far removed from the 64 kHz to which it is up-converted. Double-balanced mixers are ideal for all mixing applications. They are the first choice of mixer type if their higher cost is of no concern.

SUMMARY

The waveforms that result from intermod signals are complex and very difficult to use to determine the frequency and power levels of the intermod signals. With a spectrum analyzer, however, each of the intermod signals can be easily measured and interpreted. The spectrum analyzer indicates the frequencies of both harmonics and intermod products, sometimes totaling a dozen separate signals at significant power levels.

To keep the many intermod and harmonic signals under control, filtering and phasing networks are applied to the outputs of non-linear circuits. These networks select, reject, and/or suppress one or more of the signals created within the non-linear circuits. Phase-canceling of the input signals also removes the harmonics of these signals, along with the intermod products that result from the harmonics. The result is a far simpler and predictable distribution of frequencies with the bandwidth of the intermod products, usually only two intermod frequencies.

The applications for intermod products include modulation and mixing. Modulation is the process of applying an information signal to a higher-frequency carrier signal, to take advantages of the higher frequency's bandwidth and transmission properties. Mixing is primarily used for the frequency conversion of information signals.

The three major forms of modulation include amplitude modulation (AM), double-sideband suppressed carrier modulation (DSB-SC), and single-sideband modulation (SSB). SSB modulation is the most often used in telephone carrier communications systems, as it is the most power and bandwidth efficient of all modulation forms. Many telephone voice channels are up-converted and SSB modulated with a technique called "frequency-division multiplex" (FDM), which combines the channels for transmission on a single cable or through a single circuit.

Frequency conversion using mixers was described as a technique very much like SSB modulation, but specifically intended to convert frequencies, rather than apply information to a carrier. As with modulators, mixers are available in unbalanced and balanced forms. The simple unbalanced mixer passes all input signals, and their harmonics and intermod products, to its output. Often, the many frequencies are difficult to filter, so the unbalanced mixer is not the component of choice in critical communications circuits. The balanced mixer, like the balanced modulator, phase cancels the input signals and their harmonics, leaving only the sum and difference frequencies at its output. This mixer is extensively used within quality communications systems.

SELF-TEST REVIEW

1. Which circuit function **cannot** be used to suppress one signal, while retaining *others*, in the output waveform of a complex signal?
 - A. A linear circuit.
 - B. A filter circuit.
 - C. A phase-canceling circuit.
 - D. A balanced modulator circuit.
2. A voice signal is heard when a radio receiver is tuned to a news station. Which word identifies the frequency of the radio station?
 - A. Information.
 - B. Sideband.
 - C. Modulation
 - D. Carrier.
3. What are the two advantages of a carrier signal?
 - A. _____
 - B. _____
4. How many different *modulation forms* are likely to be used in telecommunications systems?
 - A. Four or less.
 - B. Four to six.
 - C. Seven to nine.
 - D. Ten or more.
5. In which modulation form is the *level* of the carrier signal controlled by a voice signal? _____
6. A 60 kHz signal is amplitude modulated by an 8 kHz signal. What are the frequencies of the *sideband* signals within the output waveform of the modulated signal?
 - A. _____
 - B. _____

7. A 3-volt voice signal amplitude modulates a 5-volt carrier. Calculate the *modulation index* (as a factor, and as a percentage).
- A. _____
- B. _____
8. A 2-watt carrier signal is amplitude modulated at an index of 50% (0.5). The power level of *both* sideband signals is _____ W.
9. A 2-watt carrier signal is amplitude modulated at an index of 1.0 (100%). The *total* power in the carrier and the sidebands is _____ W.
10. The same 2-watt carrier signal in Question 9 is DSB-SC modulated with the same level modulation signal. The *total* power in the carrier and the sidebands is _____ W.
11. The same 2-watt carrier signal in Question 9 is SSB modulated with the same level modulation signal. The *total* power in the carrier and the sidebands is _____ W.
12. A 20 kHz carrier signal is amplitude modulated by a 3 kHz voice signal. List the *significant frequencies* that will exist in the output waveform of the modulator, through the *third-order harmonic* and *intermod products*.
- A. _____ G. _____
- B. _____ H. _____
- C. _____ I. _____
- D. _____ J. _____
- E. _____ K. _____
- F. _____ L. _____
13. A 20 kHz carrier signal is DSB-SC modulated by a 3 kHz voice signal. List the *significant frequencies* that will exist in the output waveform of the modulator, through the third-order harmonic and intermod products.
- A. _____ D. _____
- B. _____ E. _____
- C. _____

14. Describe the major distinction between mixers and modulators.
- _____
- _____
15. A 12 kHz signal is applied to a mixer input terminal to be up converted to 67 kHz. What *two frequencies* can be applied to the other mixer input terminal to accomplish the frequency conversion?
- A. _____
- B. _____
16. A high-pass filter is to be used at the mixer output to select the 67 kHz signal in Question 15. Which of the two frequencies in your answer for Question 15 is best to use? Why?
- _____
- _____
17. Two signals are applied at the input of a mixer. Signal A is modulated on a 58 kHz carrier, and signal B is a clean unmodulated sine wave at 53 kHz. The output of the mixer is low-pass filtered at 10 kHz. What is the character of the signal at the filter's output?
- A. An unmodulated sine wave at 5 kHz.
- B. An unmodulated sine wave at 111 kHz.
- C. A modulation signal at 5 kHz.
- D. A modulated signal at 111 kHz.
18. A local oscillator signal at 105 kHz is applied to one input terminal of a mixer. A modulated carrier signal at 87 kHz is applied at the other input terminal of the mixer. The output of the mixer is high-pass filtered at 20 kHz. The *only* frequency that appears at the filter's output is at 192 kHz. What type of *mixer* is used?
- A. Unbalanced.
- B. Single-balanced.
- C. Double-balanced.
- D. Triple-balanced.

Section 4

**CREATING, AMPLIFYING,
AND
REPRODUCING SIGNALS**

CHAPTER 12 — Transducers in Audio and Telephone Systems

INTRODUCTION

In the previous chapters of this course, you learned about the fundamentals of audio and telephone signals, and the effects of passing these signals through electrical circuits. We will now study the *components* used to create and amplify these audio and telephone signals.

Electrical signals with varying waveforms can be created within electronic circuits. For most communication, however, electrical signals must originate from other sources, such as human voice. Since voices and other sources of sound produce acoustic signals, not electrical signals, the sound energy must be first converted to electrical form before the signals can be processed in electronic systems. The first stage in the telephone, and other communication systems, includes a component that converts the acoustic energy of sound signals to electrical signals. This component is called a “transducer.”



Once electrical signals have passed through their communication paths, they are most often converted back to their original form. Sound signals, originated as acoustic energy in communication systems, are converted back to acoustic energy after they have passed along a circuit as electrical energy. So, like the first stage in telephone systems, their last stage includes a component that converts electrical signals to acoustic signals. This component is also called a “transducer.” The transducer is a component used in audio systems to convert acoustic signals to electrical signals, and vice-versa.

Transducers are components used to convert energy from acoustic-to-electric form, or electric-to-acoustic form, in audio communication systems.

OBJECTIVES

When you complete this chapter, you will be able to:

- Identify the electrical principles upon which popular input and output transducers operate.
- Identify the acoustical and electrical properties of popular input transducers, such as microphones and telephone transmitters.
- Identify the acoustical and electrical properties of popular output transducers, such as loudspeakers and telephone receivers.
- Determine the efficiency of transducers from measured specifications, and convert dB_{SPL} to electrical signal power.
- Determine the electrical requirements of gain and power level for amplifier systems which use input and output transducers.

TRANSDUCERS

An electrical transducer is a component that converts energy, in other forms, to electrical energy; and electrical energy to energy, in other forms. In audio, sound, and telephone systems, the transducer converts acoustical signals to electrical signals, and vice-versa. Input transducers convert acoustical signals to electrical signals; while output transducers do the opposite, converting electrical signals to acoustical signals. Microphones and telephone transmitters (mouthpieces) are examples of input transducers. Loudspeakers and telephone receivers (earpieces) are examples of output transducers.

Input transducers convert sound waves to electrical signals. *Output transducers* convert electrical signals to sound waves.

Some transducers rely upon physical phenomena that affect electrical energy. You may recall that electrical energy can be produced in a number of ways. These are classified as electrochemical, electromagnetic, thermoelectric, photoelectric, and piezoelectric. Of these sources, *electromagnetic* and *piezoelectric* elements are used extensively as transducers. When they are used as input transducers, the physical vibrations of sound waves applied to these elements result in instantaneous voltages at their terminals. The voltages are proportional to the vibrating forces of the sound waves. When the electromagnetic and piezoelectric elements are used in reverse, as *output transducers*, electrical signals are applied to their terminals. The result is that the elements *vibrate* in proportion to the level and frequency of the applied signal voltages.

The dynamic and ribbon microphones are examples of popular electromagnetic input transducers; while the loudspeaker is the most common example of an electromagnetic transducer. These components rely upon the properties of magnetism and electromagnetic induction to convert vibrations to currents, and currents to vibrations. The loudspeaker and dynamic microphone are almost identical in general construction. In fact, inexpensive audio intercoms often switch loudspeakers to function as dynamic microphones for two-way communication.

Electromagnetic transducers rely upon *magnetism* and *electromagnetic induction* to convert sound waves to electrical signals, and vice-versa.

Figure 12-1A shows the general structure of the dynamic microphone, and loudspeaker, components. They incorporate a magnetic pole piece, surrounded by a moving coil that is attached to a flexible diaphragm. As a microphone, the acoustical waves vibrate the diaphragm, causing the coil to move in the pole's magnetic field. This induces proportional voltages at the coil's terminals. As a loudspeaker, the electrical signals are applied to the coil. The currents flowing through the coil produce proportional magnetic energy. The magnetic energy from the signal currents repel and attract the coil, and its attached diaphragm, over the magnetic pole piece, which results in acoustical vibrations.

Figure 12-1B shows the general structure of the ribbon microphone. This component is constructed with a flexible corrugated ribbon of conductive material (such as copper or aluminum foil) suspended within a magnetic field. The acoustical waves cause the foil to vibrate, as a conductor cutting magnetic lines of force. The resulting induced voltages at the ribbon's terminals are proportional to the applied acoustical waves. This type of microphone is also called a "velocity microphone", as its output voltage is proportional to the velocity of the vibrating ribbon.

Crystal and ceramic microphones are examples of *piezoelectric* input transducers. These elements depend upon acoustical vibrations causing slight bending of two crystal elements that are cemented together. The resulting stretching and compression of the crystal elements produces voltages across their structure due to the piezoelectric effect. These voltages are proportional to the acoustical waves that cause the crystal to vibrate.

Crystal and ceramic transducers rely upon the *piezoelectric effect* to convert sound waves to electrical signals, and vice-versa.

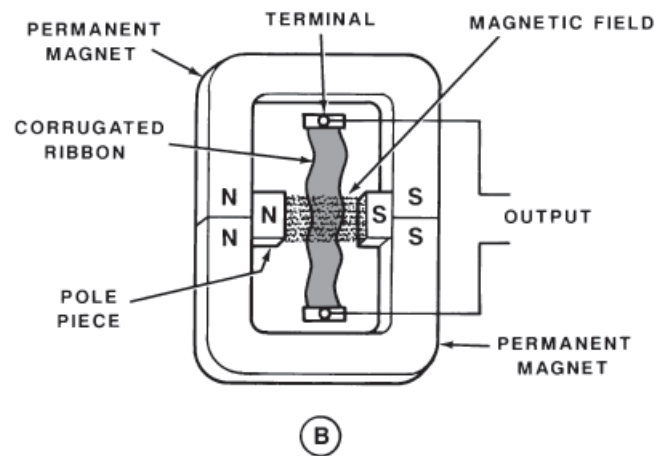
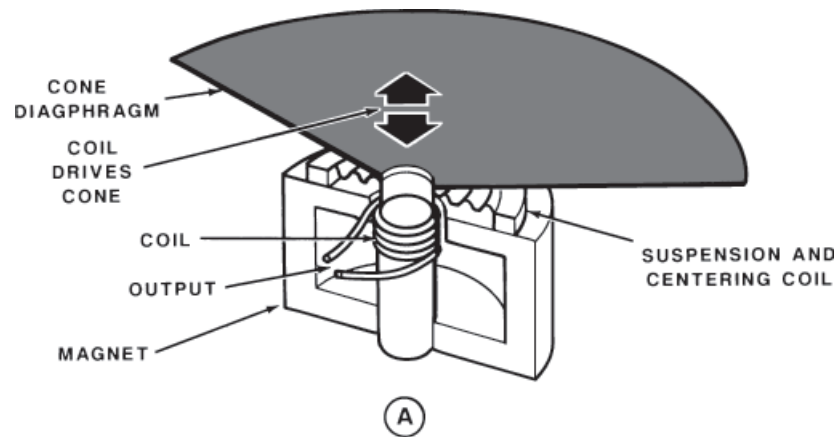


Figure 12-1
Electromagnetic microphones.

Figure 12-2 shows the general structure of the crystal and ceramic microphones. They incorporate a rigid mounting, to which the crystal or ceramic element is cemented. The diaphragm is connected to a lever arm, which applies the vibrating force of the acoustical waves to the free end of the piezoelectric element. The electrical signal voltages created by vibration appear across the terminals of the element.

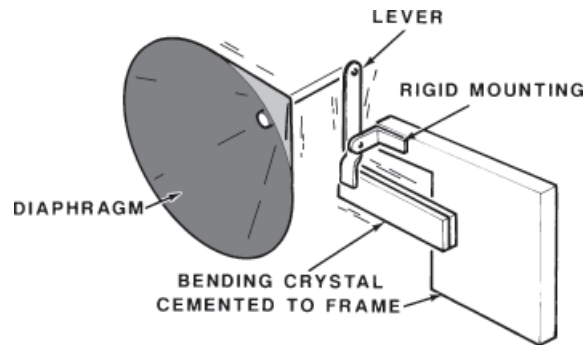


Figure 12-2

Piezoelectric microphone.

Just as the electromagnetic transducer can be reversed, so too can the piezoelectric transducer. Crystal or ceramic components are less-often used as output transducers, but they are found in headphones, earpieces, and loud-speaker “tweeters.” The structure of the output transducers are virtually identical to the microphones. Electrical signal voltages applied to the piezoelectric elements causes them to vibrate. The vibrations are coupled to the diaphragm connected to the lever arm, which creates sound waves. The crystal or ceramic output transducers are limited in their ability to vibrate at lower frequencies, so they are not used for higher-power, lower-frequency, audio applications.

Electromagnetic and piezoelectric transducers do not require any external source of power for their operation. They utilize the acoustical or electrical power applied to them to convert that form of energy. There are other types of transducer elements that do require external sources of power for their operation. The more popular of these are used as input transducers, and include the carbon microphone and the condenser microphone.

The carbon microphone is used extensively in voice communications. It is the common type of “mic” used in two-way radio, and it is most used as the transmitter of telephone sets. Unlike the electromagnetic and piezoelectric elements, the carbon mic does *not produce* electrical energy from acoustical energy. The carbon in the “mic” is a resistive element. The acoustical signals applied to the carbon mic merely *control* the current from a source of electrical power that is connected to the microphone.

Carbon microphones are transducers that utilize resistive carbon material to control current, in converting sound waves to electrical signals.

Figure 12-3 shows the structure of the carbon microphone. Carbon granules, loosely packed within a cartridge behind its diaphragm, present a moderate amount of resistance to the DC power applied across its terminals. When the diaphragm vibrates from the acoustical waves, it alternately compresses and loosens the carbon granules. The result is a varying current from the DC source, which passes through a transformer that provides an output voltage that is proportional to the acoustical signal.

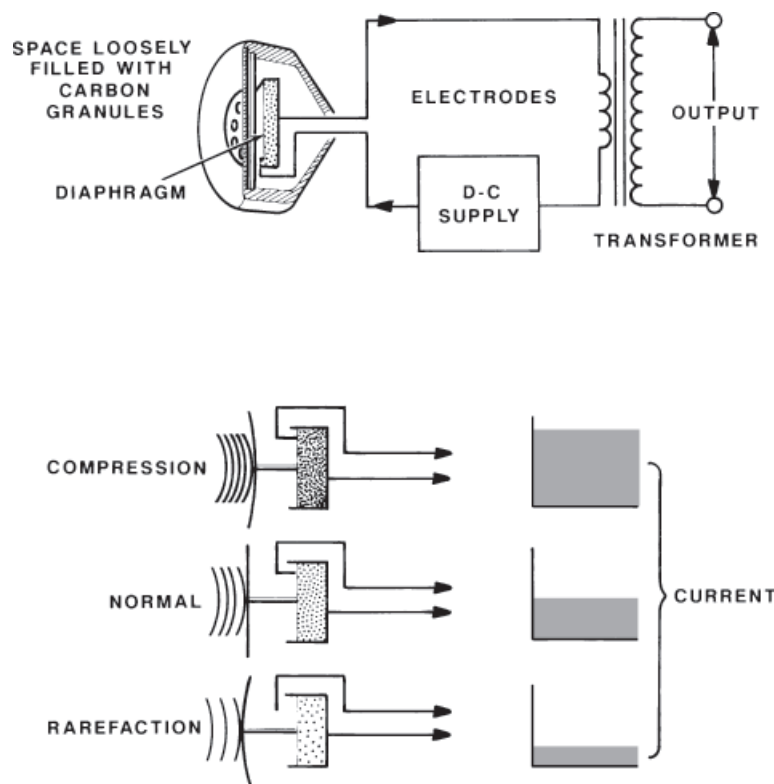


Figure 12-3
Carbon microphone.

The carbon microphone requires a source of DC *current* for operation. By contrast, the condenser microphone is an input transducer that requires a source of DC *voltage* for operation. This “mic”, sometimes called an “electret” microphone, functions as a capacitor, because it has two conductive plates separated by an insulator. One of the plates is rigid and stationary,

while the other is flexible, and serves as the microphone's diaphragm. When the acoustical waves vibrate the diaphragm, the dielectric (insulation) thickness of the capacitor changes, producing a proportional voltage across its plates.

Condenser microphones are transducers that utilize the properties of capacitance to control voltage, in converting sound waves to electrical signals.

You may recall, from your studies of electrical fundamentals, that there is a relationship between the capacity (C in Farads) and charge (Q in Coulombs) on a capacitor, and the voltage (V) across its plates. The formula is:

$$V = \frac{Q}{C}$$

With a fixed charge in DC volts across its plates, a capacitor's output voltage changes inversely with its capacitance value (C). A change in the distance between the capacitor's plates changes that capacitance, and consequently its output voltage (V).

Figure 12-4A shows the general structure of the condenser "electret" microphone. The dielectric material is metalized on one of its surfaces to form the flexible "plate" of the capacitor. The second plate of the capacitor is part of the rigid assembly structure. Studio condenser microphones have a high DC voltage applied through a power supply to "charge" the capacitor. The inexpensive electret microphones, such as those used in telephone transmitters, are charged electrostatically in the manufacturing process, and do not require an external power supply for operation. The output signal from the "mic" appears across the conductive capacitor surfaces.

The capacitor effect is sometimes used in output transducers. Figure 12-4B shows the general structure of a capacitor-type component called an "electrostatic" loudspeaker. This element has a flexible dielectric material stretched between two perforated conductive plates. A high DC voltage is applied through a power supply to charge the plates, as with condenser "mics"; and the signal is coupled to the plates through a DC blocking capacitor.

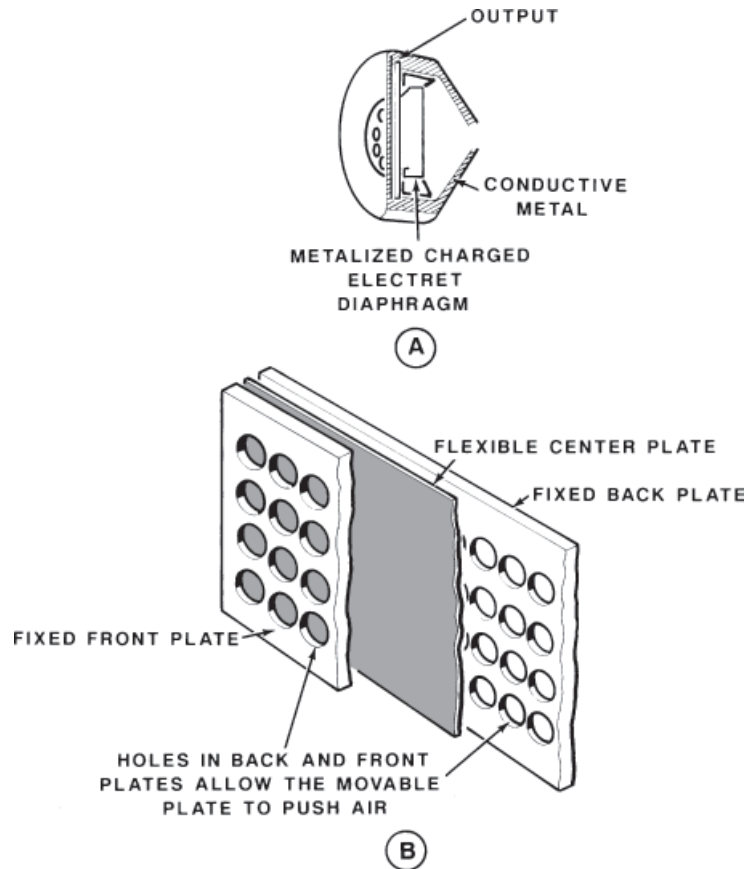


Figure 12-4
Condenser microphone.

ELECTRICAL PROPERTIES OF INPUT TRANSDUCERS

The important electrical properties of microphones include their sensitivity, frequency response, distortion, noise, and impedance. These properties determine the best use for specific microphone components within given system requirements. For example, microphones used for recording music must have wide frequency response, low distortion, and low noise properties. These features are traded-off against low sensitivity. By contrast, the transmitter microphones used in telephone sets are highly sensitive, but they have limited frequency response, distortion, and noise characteristics.

Sensitivity of Microphones

You may recall that the level of acoustical signals are measured in dB_{SPL} (sound pressure level). The standard of sensitivity is referenced to a specific sound pressure *level applied* to the microphone, with a resulting *output voltage* of the electrical signal it creates. The standard sound pressure level (SPL) is $74 \text{ dB}_{\text{SPL}}$, which is about the level of normal speech at a short distance from the microphone. The standard signal frequency is 1000 Hz. The scientific measure of the $74 \text{ dB}_{\text{SPL}}$ level is 1 microbar, which is a pressure unit of acoustic science. (Meteorologists often indicate the earth's atmospheric pressure in microbars.) For your reference, a sound pressure of 1 lb/in^2 (pound per square inch) is about $170 \text{ dB}_{\text{SPL}}$.

To fully evaluate sensitivity specifications, you should apply your knowledge of *SPL*, *dB ratios*, *referenced power*, and *power transfer* fundamentals you learned about earlier.

The specification sheets for microphones simply list dB levels for sensitivity. For example, one type of microphone is listed as having a “*sensitivity of -56 dB* .” This single number may be confusing if you aren't familiar with sensitivity specifications. A more precise specification should read:

Sensitivity: -56 dB , $0 \text{ dB} = 1 \text{ V}/\mu\text{bar}$

The interpretation of this specification is that the output voltage of the microphone is 56 dB below 1 volt, when an acoustic sound pressure level of 1 microbar (74 dB SPL) at 1 kHz is applied to the microphone's surface.

The actual voltage out of the microphone must be calculated from the voltage reference, using the formula introduced in earlier in this course. The formula is:

$$(V_{\text{out}}) = \left[\log_{-1} \left(\frac{\text{dB}}{20} \right) \right]$$

(V_{out}) is the microphone's open-circuit output voltage, and dB is its specified sensitivity. The output voltage can be calculated for this example as:

$$(V_{\text{out}}) = \left[\log_{-1} \left(\frac{\text{dB}}{20} \right) \right]$$

$$(V_{\text{out}}) = \left[\log_{-1} \left(\frac{-56}{20} \right) \right] = 1.59 \text{ mV}$$

A microphone's *sensitivity* indicates its output voltage in dB below 1 volt, when a specified level of acoustic pressure is applied to its input.

Table 12-1 includes typical sensitivity specifications for three types of microphones.

Table 12-1
Microphone specifications.

	Dynamic	Condenser	Ceramic
Frequency Response	40-12000	20-20000	40-15000
Output Impedance	600Ω	250Ω	5000Ω
Sensitivity 0 dB = 1VμBar	-50dB	-56dB	-36dB
Signal-to-Noise Ratio	60db	66dB	60dB
Maximum Input dB _{SPL}	116dB	126dB	130dB
Dynamic Range	60dB	100dB	90dB

The “transmitters” in telephone sets (“telset transmitters”) are really microphone components. Some modern lightweight telsets use electret condenser microphones as transmitters. These elements have sensitivity specifications similar to the microphones in Table 12-1. Some telsets may even use dynamic transmitters, however, *most* of the world's telephone sets use carbon microphones as transmitters. Carbon mics control the current from a power source, so they have much higher output voltage levels compared to dynamic, crystal, and condenser mics.

The sensitivity specification for the typical carbon mic is about 0 dB, with a reference of 10 microbars. 10 microbars of pressure is specified, because the transmitter is held close to the mouth of the person speaking into it. This sensitivity is about 40 dB greater than the condenser and dynamic microphones; providing an output level typically greater than 1 volt. With such a high output, the signals from carbon transmitters *don't* need amplification in the circuit between the handset and central office.

Frequency Response

The design frequency response of microphones depends upon the specific use for which the components are intended. Microphones used for musical recording have wide frequency response throughout the audio range, ideally from 20 Hz to 20 kHz. Microphones used in FM radio broadcast studios require a response 50 Hz to 15 kHz, as these frequencies are the assigned limits of FM broadcast standards. By contrast, the microphones used as telephone transmitters, *and* those used for voice communication in two-way radio, have a much narrower range of frequency response. The ideal range of frequency response for efficient voice communication is 300 Hz to 3 kHz.

Identifying frequency response by frequency range only does not completely define this important property of microphones (or other components and circuits for that matter). The second dimension of power, in reference to sensitivity, must be added to fully characterize microphones. One microphone with a 10 dB variation in sensitivity, over its operating range from 300 Hz to 3 kHz, will perform poorly compared to another microphone with only 1 dB variation over the same 300 Hz to 3 kHz frequency range.

The *frequency response* of a microphone is best shown in a graph that indicates the “mic’s” sensitivity *variation* over a range of frequencies.

The best way of identifying frequency response is with a graph of sensitivity vs. frequency. Figure 12-5 compares the frequency-response graphs for typical “mics,” intended for musical recording and voice communication. As you study these graphs, you should observe the following:

1. Our sense of hearing responds to sounds logarithmically. Therefore, the graphs are logarithmically divided in frequency along their x-axis, and logarithmically indicated in power ratios (dB) along their y-axis.
2. Most graphs show a variation in dB *only*; then list the sensitivity at 1 kHz, which is the 0 dB reference for the graph. Figures 12-5A, 12-5C, and 12-5E are so dimensioned.
3. The graphs can be dimensioned directly in sensitivity or dB with a reference to 1 microBar (or 10 microbars). Figures 12-5B and 12-5D are so dimensioned.
4. Figures 12-5A and 12-5B graph the typical frequency response of a quality studio recording microphone; while Figures 12-5C, 12-5D, and 12-5E graph the less critical frequency response characteristics of a carbon “mic” used as a telephone transmitter.
5. Figure 12-5E shows the frequency response of a poor-quality carbon “mic”, with power variations of 10 dB over its frequency range. Compare this graph to that of Figure 12-5D, which shows the “flat” frequency response of a high quality carbon microphone.

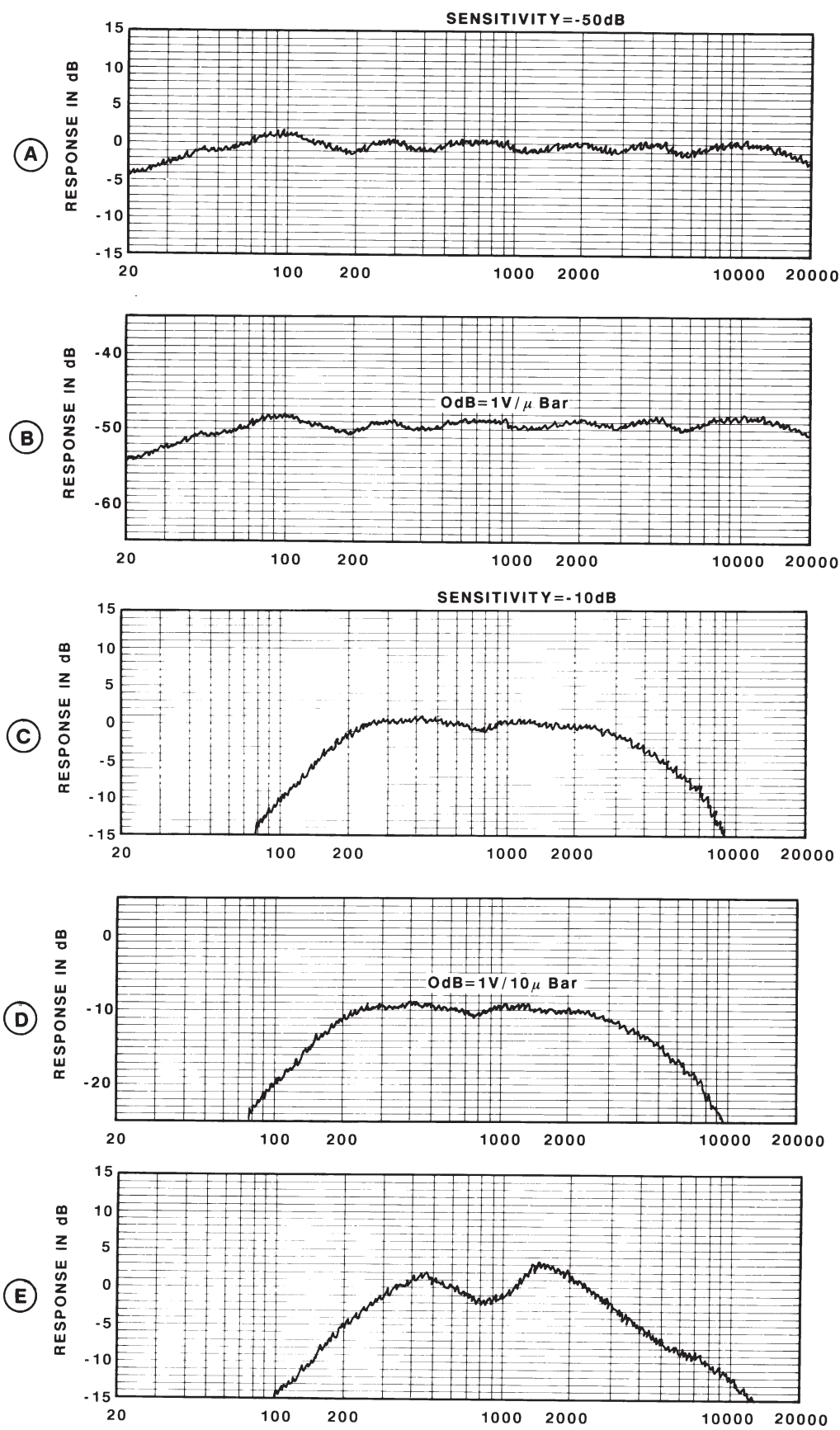


Figure 12-5
Microphone frequency response.

Noise and Distortion

As with all electrical/electronic components and circuits, there are always trade-off characteristics which must be considered when you select and use microphones. For example, one of these characteristics is apparent in the graphs of Figure 12-5. Carbon microphones are much more sensitive than condenser microphones, but they have limited frequency response performance.

Other important trade-off factors of microphone performance include distortion and noise. As with sensitivity, studio recording “mics” trade off low distortion and noise properties against sensitivity and cost. Distortion and noise are extremely critical characteristics for “mics” used for accurate recording of music, but such characteristics are less critical for microphones used in telephone and other voice transmitters.

Distortion in microphones is similar to distortion in electrical circuits that you studied earlier. Distortion results from non-linearity in the operation of the transducer components. The “mic” has some amount of harmonic distortion with each single acoustic signal applied to it, and some amount of intermodulation distortion when two or more signals are applied to it. Quality studio microphones have very low distortion factors, typically much less than 1%. However, their distortion increases if acoustic power is applied *above* a specified maximum sound pressure level (dB SPL). For most microphones, this maximum level falls between 110 and 130 dB SPL.

Harmonic and intermodulation distortion occur due to non-linearity in microphone elements. Distortion increases at higher acoustic levels.

Carbon “mics” have higher distortion factors a trade-off disadvantage of their higher sensitivity. The distortion in carbon microphones often becomes very high if the component is exposed to moisture, causing its carbon granules to become packed together. The “mic” can sometimes be restored by heating it to dry it out, and tapping it on a solid surface to loosen its granules.

Electrical noise is another undesirable property of microphones. In carbon microphones, current is continuously passing through the carbon granules. This current causes minute variations in voltage, and produces a low level of “hiss” which can be heard in the background of the signals at the microphone’s output. This “hiss” is noise. If the noise is at a high level, it will impair accurate communication by masking signals. Figure 12-6 relates the levels of signal to noise.

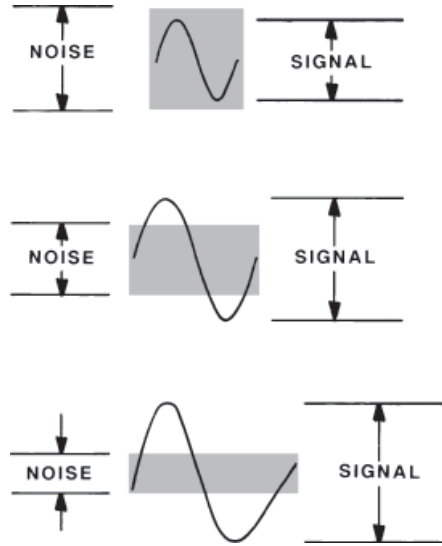


Figure 12-6

Relating signal levels to noise levels.

Electrical noise in dynamic, crystal, and condenser “mics” also originates from random electrical variations in their operation. Since these components don’t require an applied current, however, their noise levels are much lower than their carbon counterparts.

Noise is a result of random electrical variations in microphone elements. Higher levels of noise impair accurate communication by masking signals.

Under ideal conditions, the noise from the “mic” should be at least 30 dB below the signal level for voice communication, and 60 dB below the signal level for quality audio and music broadcast and recording. For telephone transmitters, the noise is limited by FCC regulation to yield 30 dB signal-to-noise ratio from a sound pressure level as low as 37 dB SPL, which is the level of very quiet speech.

Microphone Impedance and Power Transfer

A microphone is a transducer which converts acoustical energy to electrical energy. The microphone is also a signal generator, as it produces electrical signals, and is capable of transferring a specific amount of power to an electrical circuit. It is important to know the power capacity of microphones in order to assure their proper operation in electronic circuits and systems. Like all signal generators, the “mic’s” output power capacity will depend upon both its output voltage and its output (internal) impedance. You may apply the “*Power Transfer Theorem*” from Chapter 4 to microphones, as well as to electrical circuits. Using this theorem, you may conclude that the maximum power available from a microphone will be transferred to a circuit when the impedances of the microphone and circuit are matched.

The power available from a microphone depends upon its output voltage and its output impedance.

Earlier, you studied the relationship between a microphone’s sensitivity specification and its open-circuit voltage. With this calculated open-circuit voltage, and a known output impedance, the maximum power capacity of a microphone can be calculated as:

$$(P_{\max}) = \frac{(0.5 V_{oc})^2}{Z}$$

$$P_{dBm} = 10 \log P_{mW}$$

With a sensitivity of 0 dB (1-volt open-circuit), for example, a carbon “mic” with 600 ohms impedance, will have a power capacity ($P_{\max}$) of:

$$(P_{\max}) = \frac{(0.5 V_{oc})^2}{Z} = \frac{(0.5 \times 1)^2}{600} = 0.42 \text{ mW}$$

Converting that to decibels, you have:

$$P_{dBm} = 10 \log P_{mW} = 10 \log 0.42 = -3.8 \text{ dBm}$$

By contrast, the condenser studio “mic” will have a much lower power capacity. Its specifications include a sensitivity of -54 dB (2 millivolts open-circuit), and an impedance of 200 ohms. Its power capacity is calculated as:

$$(P_{\max}) = \frac{(0.5 V_{oc})^2}{Z} = \frac{(0.5 \times 2 \times 10^{-3})^2}{200}$$

$$P_{\max} = 5 \times 10^{-6} \text{ mW} = 5 \text{ nW}$$

Converting that to decibels, you have:

$$P_{dBm} = 10 \log P_{mW} = 10 \log (5 \times 10^{-6})$$

$$P_{dBm} = -53 \text{ dBm}$$

Quality microphones usually have low output impedance values. Low impedance sources have higher output power levels, and yield lower cable losses. With identical sensitivity ratings of -54 dB, for example, the 200-ohm “mic” has a power capacity of 5 nW (-53 dBm); while a 10-kohm high-impedance “mic” has a power capacity of only 0.1 nW (-70 dBm). Even though the sensitivity specifications appear comparable, the power capacities of the two mics are very different. The high-impedance “mic” is truly 17 dB (-70 dBm - (-53 dBm), less efficient than the low-impedance “mic.”

Low impedance microphones have higher output power levels, and yield lower cable losses.

While electronic amplifiers can be used to boost the power levels of microphones, the signals must first be transferred to the amplifier through a cable. Cable impedances are low, typically 600 ohms for telephone, voice, and audio signals. Transformers are often used to match the “mic” impedances to that of the cable, and thereby reduce the total system losses. (In some cases, the transformers are a part of the internal network of the microphone component. In telephones, they are a part of the telset’s network.) Keep in mind, however, that a 10-kilohm to 600-ohm transformer will step down the applied voltage, as well as the impedance. Matched impedances will only eliminate mismatch losses, as shown in Figure 12-7.

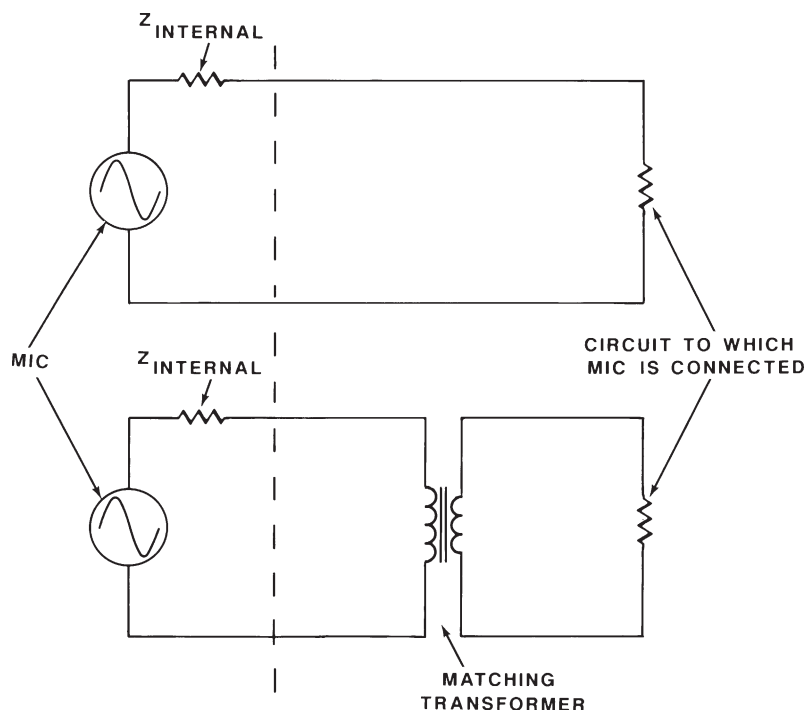


Figure 12-7

We can compare the voltage levels of the two microphones of our previous examples, presuming they are connected to matching transformers. For the 200-ohm “mic,” its output voltage from the matching transformer is calculated as:

$$(V_{sec}) = V_{pri} \sqrt{Z_{sec} / Z_{pri}} = \sqrt{600 / 200} = 1.73 \text{ mV}$$

For the 10-kilohm “mic,” its output voltage from the matching transformer is calculated as:

$$(V_{\text{sec}}) = V_{\text{pri}} \sqrt{Z_{\text{sec}} / Z_{\text{pri}}} = 1 \sqrt{600 / 10\text{k}} = 0.24 \text{ mV}$$

A 200:600-ohm matching transformer will raise the voltage level of the 200-ohm “mic” from 1 millivolt to 1.73 millivolt; while the 10k:600-ohm matching transformer will reduce the voltage level of the 10-kilohm “mic” from 1 millivolt to 0.24 millivolt. The difference in voltage from the two mics can be calculated in dB, as:

$$\text{dB} = 20 \log (V_1 \div V_2) = 20 \log (1.73 \div 0.24) = 17 \text{ dB}$$

As you may observe, the difference in signal levels from the two “mics” after transformation, is the **true power levels** of the “mics.”

ELECTRICAL PROPERTIES OF OUTPUT TRANSDUCERS

Output transducers are used extensively in electronic communication systems. They are used to convert electrical signals into acoustic waves, which we interpret as sound information. The most popular output transducers include loudspeakers, headphones, and telephone receivers. Almost all of these components are electromagnetic, “dynamic” transducers. The important electrical properties of these transducers include their efficiency (like microphone sensitivity), frequency response, distortion, and impedance. These properties are similar to those of the microphones described earlier.

Efficiency

The efficiency of output transducers is a measure of their ability to convert electrical energy to acoustical energy. Highly efficient components produce a higher level of sound from lower signal current than less efficient components. Loudspeaker efficiency is rated by measuring the sound pressure level produced when 1 watt of electrical power at 1000 Hz is applied to the speaker’s input. The measurement is made with a standard microphone of known sensitivity, placed at a distance of 1 meter from the speaker’s output surface. Figure 12-8A shows this measurement set-up.

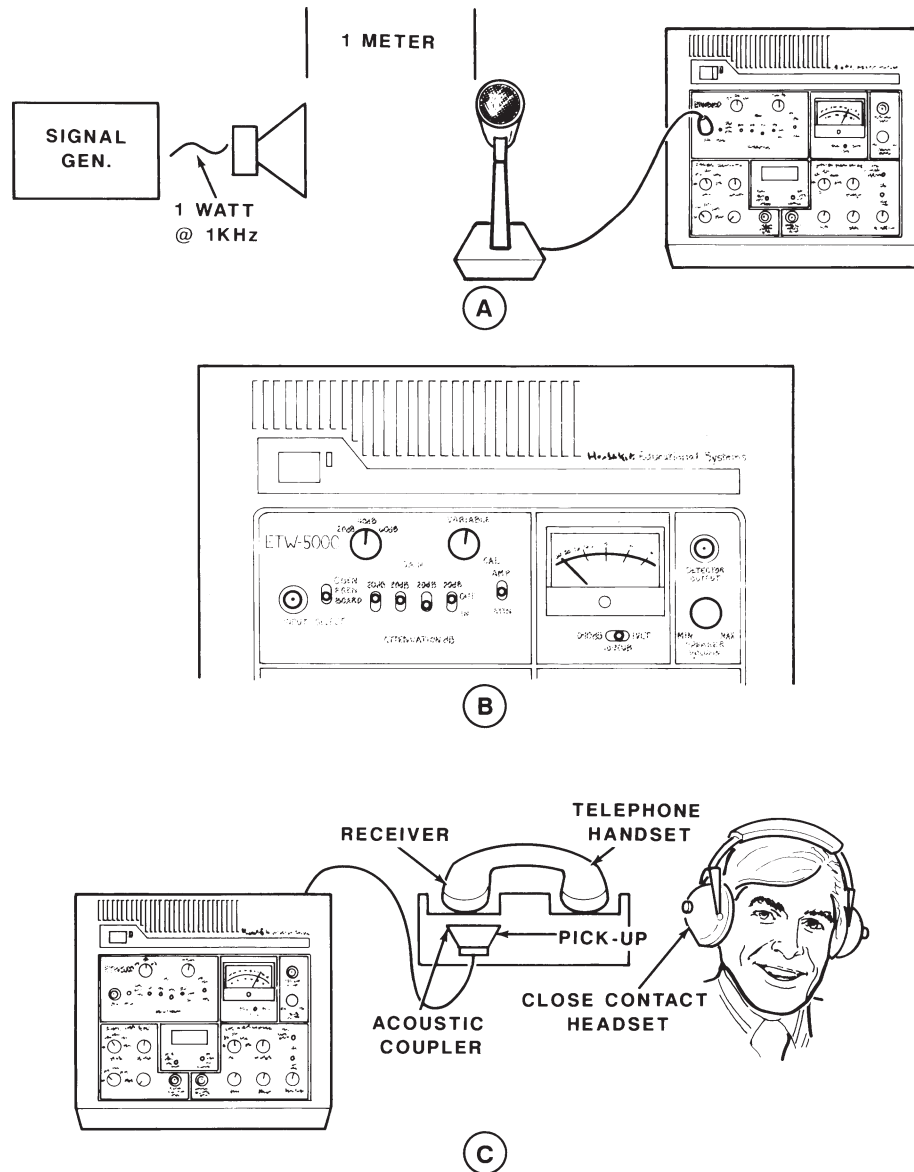


Figure 12-8
Efficiency measurement of output transducers.

Loudspeaker efficiency defines the sound pressure level produced when 1 watt of electrical power at 1000 Hz is applied to a loudspeaker's input.

A standardized microphone of known sensitivity is what we use to measure speaker efficiency. Recall that the open-circuit signal voltage from a microphone can be calculated, based upon its sensitivity specification. Earlier, we determined a “mic’s” sensitivity, with the formula:

$$V_{\text{out}} = \log_{-1} \left(\frac{\text{dB}}{20} \right)$$

For example, a microphone rated at –54 dB, referenced to 0 dB at 1 volt-per-microbar, will provide the following reference levels for 74 dB SPL:

1. A millivolt rms signal to a high impedance voltmeter.
2. A 1-millivolt rms signal into a voltmeter with impedance matched to that of the “mic.”
3. A 5.66-millivolt p-p signal directly into an oscilloscope.
4. A 2.83-millivolt p-p signal into an oscilloscope with its input loaded to match the impedance of the “mic.”

The measurement of speaker efficiency is most practically made with rms voltmeters, calibrated in dB, used in conjunction with switch attenuation in dB as shown in Figure 12-8B. Efficiency (n) is calculated from measured data with the following formula:

$$n [\text{in dB SPL/watt}] = 74 \text{ dB SPL} + \text{dB}_{\text{meter}}$$

With 1 watt applied to the loudspeaker, and 16 dB indicated on the meter and its attenuator (combined dB values), the efficiency of the speaker is:

$$n = 74 \text{ dB SPL} + \text{dB}_{\text{meter}}$$

$$n = 74 \text{ dB SPL} + 16 \text{ dB} = 90 \text{ dB SPL/watt}$$

Once the loudspeaker’s efficiency is known, it is practical to determine the applied electrical power to obtain any sound power level. The applied electrical power requirement can be calculated with the following formula:

$$P = \log_{-1} \left(\frac{\text{dB SPL} - n}{10} \right)$$

If a sound level of 104 dB SPL is required from a speaker with an efficiency of 90 dB SPL/watt, for example, the electrical signal power required can be calculated as:

$$P = \log_{10} \left(\frac{\text{dB SPL} - n}{10} \right)$$

$$P = \log_{10} \left(\frac{104 - 90}{10} \right) \approx 25 \text{ watts}$$

The measurement of efficiency for headphones and telephone receivers differs from loudspeaker measurements in both the distance between the standard microphone and the output transducer, and in the level of power applied to the transducer. For this measurement, the two devices are in close contact with each other, just as the human ear will be in contact with the headphone or receiver. This set-up is shown in Figure 12-8C.

For these contact efficiency measurements, the “mic” reference level will still be 1 microbar at 74 dB SPL, but the level of the applied electrical signal will be reduced by 30 dB, to a more practical 1.0 milliwatt. Except for these two differences, the techniques of calculating efficiency are identical. However, the reference efficiency will still be indicated in units of dB SPL/watt, by factoring in the 30 dB difference in test signal level in your calculations.

For example, a microphone with a sensitivity of –54 dB is placed in contact with a headphone transducer driven by a 100 milliwatt, 1000 Hz signal. The output from the “mic” reads 2 millivolts rms on a high-impedance voltmeter when the output signal level is reduced by 12 dB. The efficiency of the transducer is calculated as:

$$n [\text{in dB SPL/watt}] = 74 \text{ dB SPL} + \text{dB}_{\text{meter}} + 30 \text{ dB}$$

The last 30 dB in the formula accounts for the 1.0 milliwatt (instead of 1 watt) signal applied to the transducer. With 1 milliwatt applied to the headphone, and 12 dB indicated on the meter and its attenuator, the efficiency of the headphone is:

$$n = 74 \text{ dB SPL} + \text{dB}_{\text{meter}} + 30 \text{ dB}$$

$$= 74 \text{ dB SPL} + 12 \text{ dB} + 30 \text{ dB} = 116 \text{ dB SPL/watt}$$

Keep in mind that this specification of a high dB SPL/watt is based upon the headphone being in contact with the ear or “mic.” To make a comparison with a loudspeaker, where the “mic” is placed at 1 meter from the transducer, the sensitivity factor will be reduced by 30 dB.

Once the efficiency of a transducer is known, its electrical operating requirements can be determined, just as with the loudspeaker. For example, if a sound level of 80 dB SPL is required from a telephone receiver which has an efficiency of 112 dB SPL/watt, the required driving power level can be calculated as:

$$P = \log_{-1} \left(\frac{\text{dB SPL} - n}{10} \right) = \log_{-1} \left(\frac{80 - 112}{10} \right)$$

$$P = 0.6 \text{ mW}$$

The electrical signal power required to drive the telephone receiver of our example to a level of 80 dB SPL is 0.6 milliwatt (−2.2 dBm).

Frequency Response

The frequency standard for acoustical measurements is 1000 Hz, so too is the measurement standard for input and output transducers. How far above and below 1000 Hz the transducers efficiently produce sound signals depends upon their application. For telephone systems, the receiver transducers operate over a frequency range that is limited to voice communication. For the accurate reproduction of music, loudspeakers and headphones operate over an extended frequency range to the limits of human hearing.

As we described earlier for microphones, frequency response is a two-dimensional property, comparing frequency and power. For loudspeakers, headphones, and telephone receivers, a frequency response graph, that is referenced to efficiency, will most accurately detail their operation.

Figure 12-9 shows a series of representative frequency response graphs that detail the variation in output transducer power over their frequency range of operation. The graphs are logarithmically scaled in dB vs. frequency, referenced to a 1000 Hz efficiency factor, and measured in dB SPL/watt.

1. Figure 12-9A and 12-9B graph the response of a telephone receiver. Observe that the Y-axis of Figure 12-9A is scaled in dB variation; while the Y-axis of Figure 12-9B is scaled in dB SPL.
2. The frequency response indicated in Figures 12-9A and 12-9B shows a 5 dB variation from 300 Hz to 5 kHz, and a 15 dB variation from 200 Hz to 10 kHz.
3. Figure 12-9C shows the frequency response of a large efficient loudspeaker which has good low-frequency response, down 10 dB at about 60 Hz. It has a 7 dB peak at 500 Hz. Its response then drops above 4 kHz.

4. Figure 12-9D shows the frequency response of a small inefficient loudspeaker. However, its frequency response is quite good, with only a 6 dB variation in response from about 100 Hz through 16 kHz. Its efficiency at 1 kHz is only about 76 dB SPL.

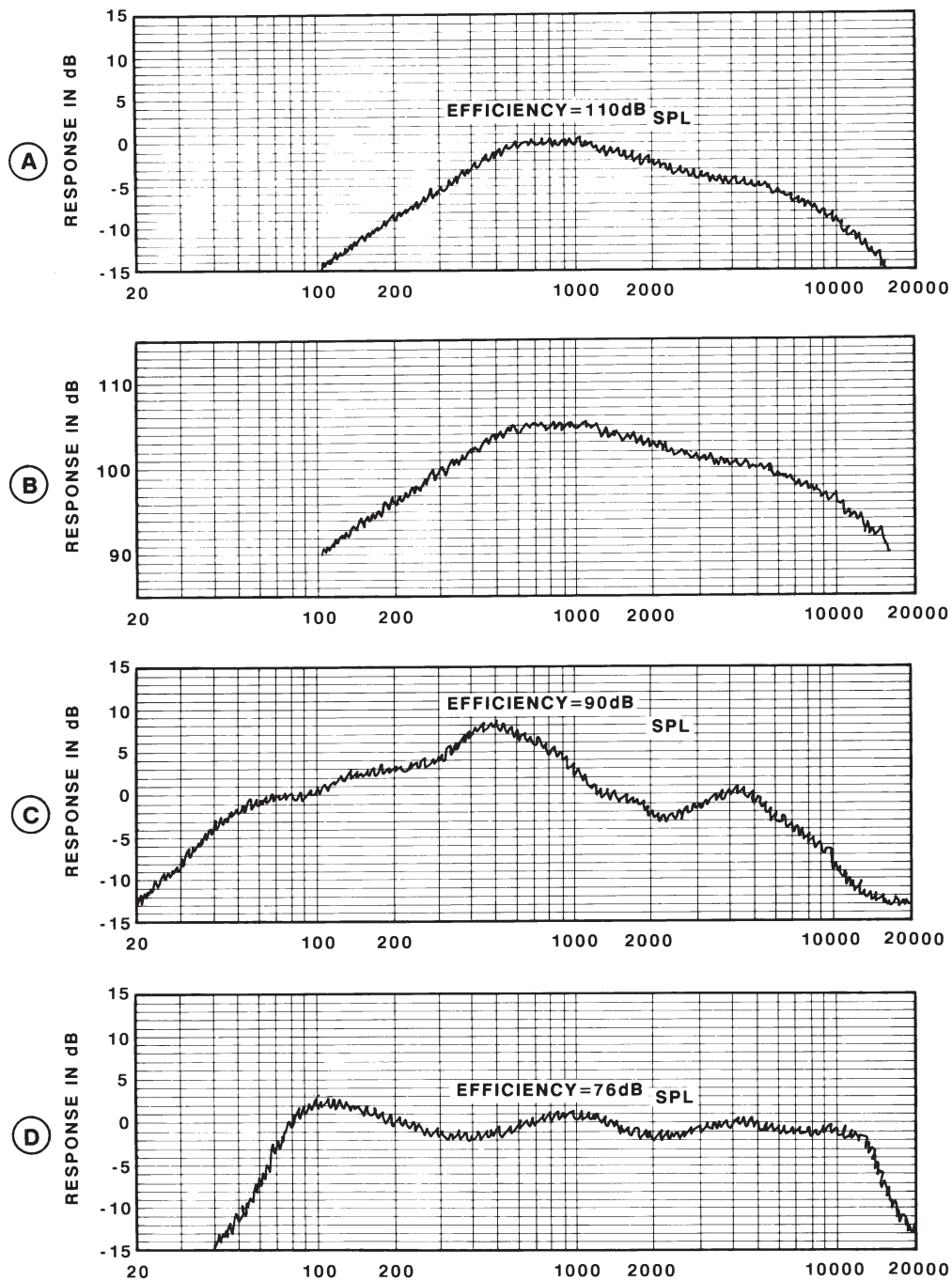


Figure 12-9
Frequency response of output transducers.

As you may conclude from your study of the graphs of Figure 12-9, output transducers do have distinctive frequency response characteristics. These are the properties that make loudspeakers, headphones, and telephone receivers sound different to our ears. The output transducers that are more efficient will produce a higher level sound with the same applied signal power, than those that are less efficient. There is a “trade-off”, however, as it is easier to design and manufacture less efficient transducers that respond uniformly over a wider range of frequencies. Efficient loudspeakers, with wide frequency response, are very difficult and costly to manufacture.

Impedance, Power, and Distortion

The amount of signal current that flows through a dynamic (electromagnetic) transducer, and the power it consumes, depends upon the voltage level of the applied signal and the internal impedance of the transducer. Loudspeakers require high magnetic energy to move a large amount of air in the sound waves they produce; so they require high levels of signal current to drive them.

It is correct to conclude that loudspeakers should be low-impedance components that consume higher levels of power. The typical loudspeaker impedance is 8 ohms. By contrast, telephone receivers need not move large amounts of air, as they are in close contact with the ears of listeners. Telephone receivers are thus high-impedance components that operate at low power levels. The telephone receiver is often a transformer coupled to match a standard 600-ohm line impedance. The impedance of the receiver itself is typically in the range of 300 ohms.

With either low- or high-impedance transducers, there is a limit to the power they can process. At excess power levels, the elements within the transducers will approach their physical limits and become non linear. Recall, that non-linear components produce distortion in the signals they process. When the power levels of the signals applied to the transducers approach the limits of their design, distortion increases rapidly. Three percent harmonic distortion is the generally accepted limit for loudspeakers, headphones, and telephone receivers.

The power limits of transducers are designed at the level where they become non-linear and distort the waveforms of the signals applied to them.

Table 12-2 compares the distortion versus output power of two loudspeakers. Note that 3% (2.5% 2nd harmonic + 0.5% 3rd harmonic) distortion factors for the larger speaker occur at 115 dB SPL; while 3% (2.1% 2nd harmonic + 0.9% 3rd harmonic) for the smaller speaker occur at 97 dB SPL. It is clear that the larger speaker has a higher power limit, and lower distortion at higher power levels. But look more closely at low-power operation of the two speakers, in the 75 to 85 dB SPL range, and note that the harmonic distortion of the smaller speaker is about half that of the larger speaker. There is a “trade-off” in distortion factors for loudspeakers, just like their power factors. It is easier to design and manufacture less efficient transducers with low distortion, than it is to design highly efficient transducers with low distortion.

Table 12-2
% Loudspeaker Distortion.

Output dB SPL	Large Speaker		Small Speaker	
	2nd-har.	3rd-har.	2nd-har.	3rd-har.
75	0.6	0.5	0.4	0.2
85	0.6	0.5	0.3	0.2
95	0.7	0.6	1.4	0.8
97	1.1	0.7	2.1	0.9
105	1.2	0.6		
115	2.5	0.5		

The applied signal level that produces distortion is usually specified as the upper power limit of a transducer’s dynamic range of operation. (This level is sometimes exaggerated in sales jargon.) This level can be calculated based upon the efficiency of the speaker. The procedures are identical to those used earlier in determining the electrical power necessary to produce a given sound power level:

$$P = \log_{-1} \left(\frac{\text{dB SPL} - n}{10} \right)$$

Assuming the efficiency (n) of the larger speaker of our example is 94 dB SPL, and its 3% distortion level is 115 dB SPL, its maximum applied power is calculated as:

$$P = \log_{-1} \left(\frac{\text{dB SPL} - n}{10} \right)$$

$$P = \log_{-1} \left(\frac{115 - 94}{10} \right) = 125 \text{ watts}$$

Assuming the efficiency (n) of the smaller speaker of our example is 76 dB SPL, and its 3% distortion level is 97 dB SPL, its maximum applied power is calculated as:

$$P = \log_{-1} \left(\frac{\text{dB SPL} - n}{10} \right)$$

$$P = \log_{-1} \left(\frac{97 - 76}{10} \right) = 125 \text{ watts}$$

From these calculations, you should conclude that output transducer power limits depend upon the *difference* between their 3% distortion level, and their efficiency, not the absolute values of these factors.

SUMMARY

In this chapter we reviewed the types and properties of transducers. We identified input transducers as those components that convert sound waves to electrical signals. The popular input transducers that are used in audio and telecommunication systems are microphones and telephone transmitters. We also identified output transducers as those components that convert electrical signals to sound waves. The popular output transducers used in audio and telecommunication systems include loudspeakers, headphones, and telephone receivers.

Among the most important specifications of input transducers are their sensitivity and impedance. Sensitivity quantifies the signal voltage level at a microphone's output, with a given sound level applied to its input. High sensitivity mics require less signal amplification than those of low sensitivity. Impedance indirectly quantifies the true power capacity of a microphone. Low-impedance mics have far greater capacity than high-impedance mics.

The other important specifications of input transducers include noise and distortion. Noise reduces sensitivity and accuracy of communication in systems. The carbon mic used in telephone transmitters is the most noisy of designs. Distortion usually occurs in microphones when they are driven above their input level limits. These limits are specified for microphone performance.

Output transducers must move sufficient amounts of air to be heard as sound waves. Loudspeakers are transducers that require a substantial level of power to produce sounds that can be heard at some distance by the ear, while headsets and telephone receivers are transducers that operate at significantly lower power levels, as they are in contact with the ear. Loudspeakers, headsets, and telephone receivers are specified by their output power efficiency, using the same units of measurement. The difference in efficiency between the two types of devices' output levels are compensated by factoring 30 dB into calculations.

The power capacity of output transducers depends upon their impedance and their efficiency. The power limits of these components are usually specified for output sound levels that reach 3% harmonic distortion. Efficient speakers reach their levels of distortion at higher sound power levels than speakers of lower efficiency. The factors of output transducer efficiency and distortion determine the amount of electrical power required to drive them, and the degree of signal amplification required in communication systems.

The frequency response of both input and output transducers is a matter of design trade-offs against efficiency, sensitivity, and cost. Extended frequency response is not a requirement for telephone systems, consequently the transducers (transmitters and receivers) in telephone systems are high in sensitivity and efficiency. Conversely, wide frequency response microphones have low sensitivity, and wide frequency response loudspeakers are generally lower-efficiency components.

SELF-TEST REVIEW

1. What is the physical phenomena under which ribbon microphones are classified?
 - A. Electromagnetic.
 - B. Photoelectric.
 - C. Piezoelectric.
 - D. Electrochemical.

2. What is the physical phenomena under which ceramic microphones are classified?
 - A. Electromagnetic.
 - B. Photoelectric.
 - C. Piezoelectric.
 - D. Electrochemical.
3. Which of the following microphones require an external source of current for operation?
 - A. Dynamic.
 - B. Crystal.
 - C. Condenser.
 - D. Carbon.
4. Which of the following microphones require an external source of voltage for operation?
 - A. Dynamic.
 - B. Crystal.
 - C. Condenser.
 - D. Carbon.
5. Which of the following microphones have the highest sensitivity characteristics?
 - A. Dynamic.
 - B. Crystal.
 - C. Condenser.
 - D. Carbon.
6. Which of the following microphones are most often used as telephone transmitters?
 - A. Dynamic.
 - B. Crystal.
 - C. Condenser.
 - D. Carbon.
7. A reference sound pressure level equal to 10 microbars is:
 - A. 64 dB SPL.
 - B. 74 dB SPL.
 - C. 84 dB SPL.
 - D. 94 dB SPL.

8. A microphone's output is listed as -40 dB, referenced to 0 dB = $1\text{ V}/\text{microbar}$. What is its open-circuit output voltage when a sound pressure level of 74 dB SPL is applied to its input?
- A. 1 mV.
 - B. 3.16 mV.
 - C. 10 mV.
 - D. 31.6 mV.
9. A microphone's output is listed as -40 dB, referenced to 0 dB = $1\text{ V}/\text{microbar}$. What is its open-circuit output voltage when a sound pressure level of 64 dB SPL is applied to its input?
- A. 1 mV.
 - B. 3.16 mV.
 - C. 10 mV.
 - D. 31.6 mV.
10. A microphone's output is listed as -40 dB, referenced to 0 dB = $1\text{ V}/10$ microbar. What is its open-circuit output voltage when a sound pressure level of 64 dB SPL is applied to its input?
- A. 1 mV.
 - B. 3.16 mV.
 - C. 10 mV.
 - D. 31.6 mV.
11. Sensitivity specifications for transducers are referenced to a frequency:
- A. of 300 Hz.
 - B. of 1 kHz.
 - C. of 3 kHz.
 - D. that varies depending upon the application.
12. Frequency response specifications for transducers compare two dimensions of signals. These are:
- A. frequency and power.
 - B. frequency and time.
 - C. frequency and voltage.
 - D. frequency and current.

13. Which of the following microphone types produces the highest levels of electrical noise?
- A. Dynamic.
 - B. Crystal.
 - C. Condenser.
 - D. Carbon.
14. The higher power limits of transducers occur at signal levels that cause:
- A. 1% Distortion.
 - B. 3% Distortion.
 - C. 10% Distortion.
 - D. burn-out.
15. For maximum power transfer from a microphone to a circuit, the impedance of the circuit should be:
- A. half the microphone impedance.
 - B. equal to the microphone impedance.
 - C. twice the microphone impedance.
 - D. ten times the microphone impedance.
16. When transformers are used to change transducer impedances, their input-to-output voltage ratios are equal to:
- A. the square root of their impedance ratios.
 - B. their impedance ratios.
 - C. the square of their impedance ratios.
 - D. ten times their impedance ratios.
17. Two microphones are rated at the same sensitivity, but one has 200 ohms impedance, while the other has 600 ohms impedance. Answer 200 or 600 to the following questions:
- A. Which "mic" has a higher power capacity? _____
 - B. Which "mic" will have a higher output voltage when driving a 600 ohm circuit? _____
 - C. Which "mic" will have a higher output voltage when driving a 200 ohm circuit? _____

18. Loudspeaker efficiency is a measure of the sound pressure level that results from driving the speaker with an electrical input power of:
- A. 1 milliwatt.
 - B. 100 milliwatts.
 - C. 1 watt.
 - D. 10 watts.
19. When measuring the efficiency of headphones and telephone receivers, what correction value is applied to account for the close contact between the driver transducer and pick-up "mic"?
- A. 0 dB.
 - B. 10 dB.
 - C. 20 dB.
 - D. 30 dB.
20. A large loudspeaker is 20 dB more efficient than a small loudspeaker. The small speaker reaches its 3% distortion level at 30 dB above its efficiency value (n); while the large speaker reaches its 3% distortion level at 26 dB above its efficiency value. Which speaker will have the higher power rating, large or small? _____
21. A dynamic microphone is specified for sensitivity at -53 dB, with 0 dB = 1 Volt/microbar. Calculate its open-circuit output voltage level.

22. A condenser "mic" is specified for sensitivity at -53 dB, with 0 dB = 1 Volt/10 microbars. Calculate its open-circuit output voltage level.

23. A sound pressure of 84 dB SPL is applied to both the dynamic and condenser "mics" in question 21 and 22. Calculate the open circuit output voltage of each of the "mics".
- dynamic (21.) _____ condenser (22.) _____
24. Refer to Figure 12-5, and calculate the following:
- A. The open-circuit output voltage at 10 kHz for the condenser "mic" in Figure 12-5B. _____
 - B. The open-circuit output voltage at 400 Hz for the carbon "mic" in Figure 12-5D. _____
 - C. The variation in dB from 300 Hz to 3 kHz for the carbon "mic" in Figure 12-5D. _____

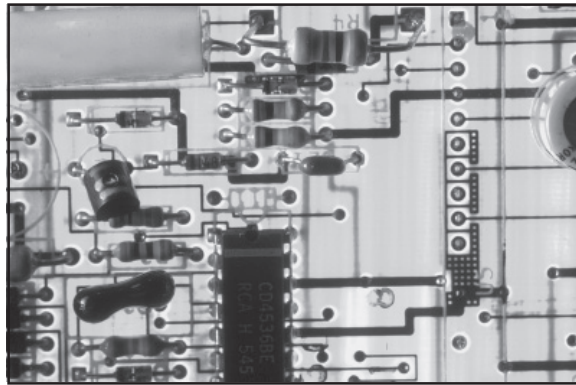
25. A high-output carbon mic has a measured sensitivity of 0 dB, with a reference of 1 V/10 microbars (84 dB SPL). In operation, its noise level is measured to be 30 dB below quiet speech at 44 dB SPL. Calculate the "mic's" open-circuit noise output voltage. _____
26. Two microphones are rated at the same sensitivity, -50 dB. The impedance of one "mic" is 200 ohms, while the impedance of the other is 5 kilohms. Calculate the power capacity of each of the "mics" (driving matched impedance circuits). Answer in mW and dBm.
- A. (200 ohm "mic") $P =$ _____ mW _____ dBm
B. (5 kilohm "mic") $P =$ _____ mW _____ dBm
27. The 200-ohm "mic" in Question 26, is transformer coupled to match it to a 600-ohm circuit. The signal voltage at the matched 200-ohm transformer primary winding is 2 mV. Calculate the level of the signal voltage on the 600-ohm circuit driven by the transformer's secondary winding. _____
28. A test circuit is connected per Figure 12-8 to measure the efficiency of a loudspeaker. The output from the standard microphone reads 2 mV on a high impedance voltmeter with 20 dB set into the meter's attenuator. Calculate the efficiency of the speaker, in dB SPL/watt. _____
29. Calculate the electrical signal power level required to drive the speaker in Question 28, above, to a sound level of 98 dB SPL. _____
30. A test circuit is connected to measure the efficiency of a telephone receiver. The output from the standard microphone in contact with the receiver, reads 2 mV on a high impedance voltmeter with 13 dB set into the meter's attenuator. Calculate the receiver's efficiency, in dB SPL/watt. _____
31. Calculate the electrical signal power level required to drive the receiver in question 30, to a sound level of 78 dB SPL. _____
32. A loudspeaker with an efficiency of 82 dB SPL reaches 3% distortion when it is driven to 105 dB SPL. Calculate the maximum power rating of the speaker in watts. _____

CHAPTER 13 — Audio Voltage and Power Amplifiers

INTRODUCTION

Amplifiers are used for many functions in electronic communication systems. At this time, we are concerned with amplifier functions that process electrical signals from input transducers, and those that apply electrical signals to drive output transducers. The electrical power levels from input transducers typically fall in the microwatt range; while those required to drive loudspeakers fall in the range of tens of-watts. The amplifiers raise these low input power levels to the high levels required to drive the output transducers.

In our study of amplifiers, we will limit our attention primarily to information about their application, and not how to design them into circuits. Special courses in electronic devices and circuits do devote a great deal of attention to the operating details of amplifier circuitry, and the dimensions of the electronic components used to make the circuits function. It is the purpose of this course, however, to describe how to *use* amplifier circuits, and their electrical *properties*, but neglect the details of their circuit design.



Modern electronic amplifiers are comprised of transistors of different types, and of integrated circuit (IC) devices. Very often, amplifiers are packaged into self-contained sub-system components that require only a direct connection to the system in which they are to operate. In all cases, the amplifier circuits have common electrical properties that define the way they process the signals applied to them. You have studied most of these properties in earlier chapters of this course. These properties include:

1. Power Level (in watts, milliwatts, or dBm).
2. Power Gain (in ratios or dB).
3. Voltage Gain (in ratios or dB).
4. Frequency Response and Phase Response.

5. Noise Level and Signal-to-Noise Ratios.
6. Circuit Linearity and Distortion Factors.
7. Signal Waveforms and Harmonics.
8. Signal Bandwidth and Circuit Bandpass.
9. Creation of Intermodulation Products.
10. Frequency Selection and Filtering.

Regardless of which type of device or circuit is used to amplify audio frequency signals, some, or all, of these ten properties must be considered in its operation.

OBJECTIVES

When you complete this chapter, you will be able to:

- Identify the distinctions used to classify voltage amplifiers vs. power amplifiers in electronic circuitry.
- Identify and categorize the dominant electron devices used to amplify signals, including bipolars, FETs, and ICs.
- Identify and categorize the dominant circuit configurations in which bipolar, FET, and IC devices are placed for signal amplification.
- Determine the operating characteristics of amplifier devices and circuits, including gain, impedance, and power levels.
- Determine the effects of amplifier circuits upon the frequency response, noise, and distortion of signals passing through them.

TRANSISTOR AMPLIFIERS

You should have some knowledge of transistor device and circuit fundamentals from your study of *Concepts of Electronics*, the prerequisite for this course. Recall, that there are two classifications of transistors, bipolar and field effect (FET); with two physical types in each classification, NPN, PNP, and N-channel and P-channel; and two more popular circuit connections for each classification: common emitter, common-collector, common-source and common-drain.

Figure 13-1 shows each of the circuit connections described in the previous paragraph, arranged by the way they process signals. Figure 13-1A shows NPN bipolar and N-channel FET devices connected as common emitter and common-source amplifiers, with a positive polarity power supply. Figure 13-1B shows PNP bipolar and P channel FET devices connected as common-emitter and common source amplifiers, with a negative-polarity power supply. These amplifier connections offer moderate input and output impedances, with high voltage and current gain factors. The identifying features of these amplifier circuits are:

1. Input signals are applied to the base terminal of the bipolar transistor, or the gate terminal of the FET.
2. Output signals are available from the collector terminal of the bipolar transistor, or the drain terminal of the FET.

Figure 13-1C shows NPN bipolar and N-channel FET devices connected as common-collector and common-drain amplifiers, with a positive-polarity power supply. Figure 13-1D shows PNP bipolar and P-channel FET devices connected as common-collector and common-drain amplifiers, with a negative-polarity power supply. These amplifier connections offer high input impedance, low output impedance, with low voltage gain and high current gain factors. The identifying features of these amplifier circuits are:

1. Input signals are applied to the base terminal of the bipolar transistor, or the gate terminal of the FET.
2. Output signals are available from the emitter terminal of the bipolar transistor, or the source terminal of the FET.

The input impedance at the gate terminal of the FET is significantly higher than the input impedance at the base terminal of the bipolar transistor. The FET is defined as being “voltage controlled”, while the lower input-impedance bipolar transistor is defined as being “current controlled.” FETs are less commonly used than bipolar transistors, as they are usually required only for high-impedance circuit applications.

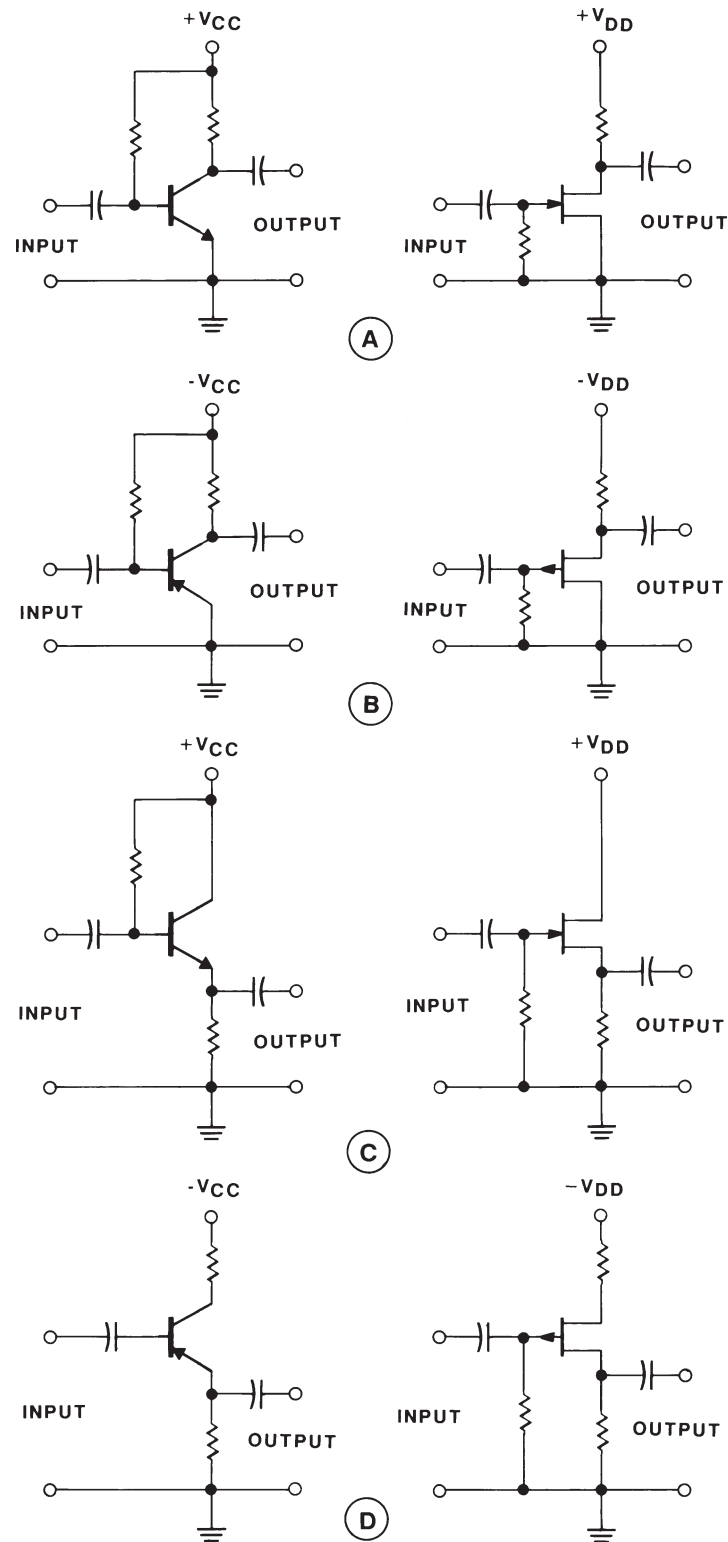


Figure 13-1
Basic transistor circuits.

The power capacities of these amplifier circuits are generally limited by their power supply voltage, and the ability of the transistors to dissipate heat. For higher power operation, the transistors are attached to large heat sinks to help dissipate the heat energy they produce. The demand for power from the circuits in which the transistors are placed, depends upon the impedance, or resistance, of the circuit or component driven by the transistor.

The power capacities of amplifier circuits are limited by their power supply voltage, and the ability of the transistors to dissipate heat.

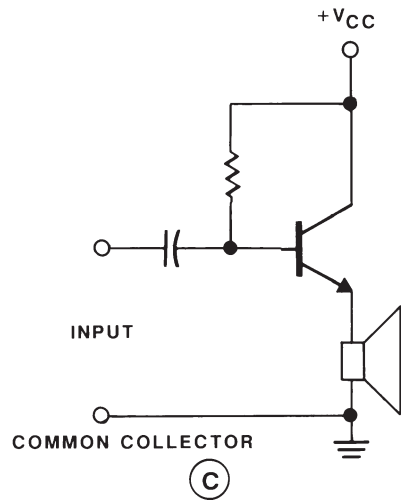
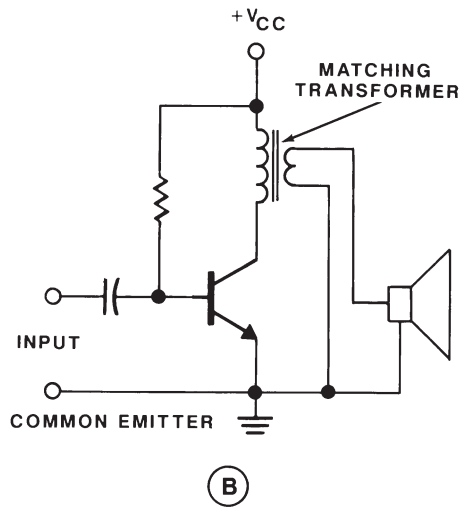
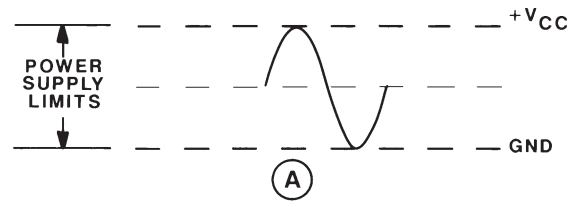
Figure 13-2 illustrates the variables of power demands and limits upon the basic transistor amplifier circuits of Figure 13-1. Figure 13-2A shows the range of peak-to-peak signal voltage “swing” available from the limits of the power supply voltage V_{CC} . Figure 13-2B shows a common-emitter amplifier circuit connected through a transformer to an 8-ohm loudspeaker as a load component. For maximum efficiency, the transformer’s primary impedance must match the transistor’s collector impedance, and the transformer’s secondary impedance must match the 8-ohm speaker impedance. The signal power limits of this circuit can be determined by with the following formula:

$$P_{MAX} = \frac{(V_{CC})^2}{8 Z_{LOAD}}$$

(This formula was introduced in Chapter 2 of this course.) For a V_{CC} of 12 volts and a collector impedance of 1000 ohms, for example, the power limit of the circuit is:

$$P_{MAX} = \frac{(V_{CC})^2}{8 Z_{LOAD}}$$
$$P_{MAX} = \frac{(12)^2}{(8 \times 1000)} = 18 \text{ mW}$$

The power is limited to a low level, because the 12-volt p-p signal “swings” across a 1000-ohm impedance. Raising the supply voltage from 12 to 24 volts will increase this power rating of 18 mW to 72 mW.

**Figure 13-2**

Transistor circuit power limits/demands.

For direct drive of the speaker, we need the low-impedance transistor connection of Figure 13-2C. This common collector circuit is sometimes called an “emitter follower”. This circuit has a low output impedance and high current gain, but no voltage gain. (The voltage gain factor is about 0.98.) With low emitter impedance with this connection, it is possible to drive the speaker directly without needing a transformer.

We can calculate the power limit of this common collector circuit for a V_{CC} of 12 volts and an emitter impedance of 8 ohms as:

$$P_{MAX} = \frac{(12)^2}{(8 \times 1000)} = 18 \text{ mW}$$

$$P_{MAX} = \frac{(12)^2}{(8 \times 8\Omega)} = 2.25 \text{ W}$$

This higher power limit results from the lower impedance output connection of the transistor. Raising the supply voltage from 12 to 24 volts will increase this power rating of 2.25 watts to 9 watts. Doubling the supply voltage increases the power by the square of two; a factor of four.

COMBINING AMPLIFIER CIRCUITS

The trade-offs of the individual circuit power limits can be used to advantage by combining circuits. The common-emitter circuit of Figure 13-2B has high voltage gain; but it has high collector impedance, which limits its power capacity. The common-collector circuit of Figure 13-2C has a high power capacity, because its emitter impedance is low; but it has no voltage gain. Figure 13-3 shows these two circuits connected together to take advantage of each of their superior properties. When so connected, the combined circuits provide both high gain and high power capacity.

Circuits of different design are combined into multi-stage circuits to take advantage of the superior properties of the individual circuits.

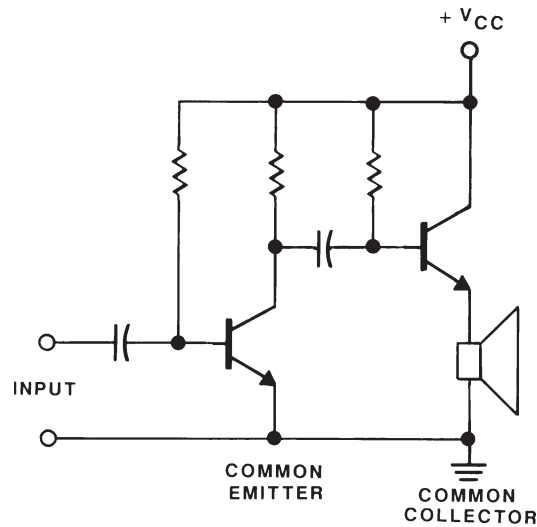


Figure 13-3
Combined power amplifier circuit.

There is yet another disadvantage in power capacity when the common-collector connection is used for low-impedance components. DC “bias” current continuously flows through the transistor, regardless of the signal level. This is a disadvantage in power capacity, as the transistor must dissipate not only the power from the applied signal, but also the continuous DC power from the power supply.

We can calculate the DC power for the common-collector circuits of Figure 13-2C and 13-3. The current through the transistor will be half the supply voltage divided by the load impedance, and the DC power is the product of that voltage times that current.

$$P_{DC} = V \left(\frac{V}{2 Z_{LOAD}} \right) = \frac{(V_{CC})^2}{2 Z_{LOAD}}$$

With a supply voltage (V_{CC}) of 12 VDC, and the 8-ohm load impedance, the DC power dissipation of the transistor is:

$$P_{DC} = V \left(\frac{V}{2 Z_{LOAD}} \right) = \frac{(12)^2}{2 \times 8\Omega} = 9 \text{ W}$$

The DC power dissipated is four times greater than the signal power! This is a poor power efficiency of only 20%.

The circuit of Figure 13-4 replaces the single-transistor common collector circuit with two transistors that operate with much better signal-to-DC power efficiency. This circuit is called a “complementary-symmetry” connection, as it uses symmetrical NPN and PNP transistors. Figure 13-4A is the complementary-symmetry circuit alone; and Figure 13-4B has this circuit connected to the common-emitter voltage amplifier as a driver, similar to Figure 13-3.

The individual NPN and PNP transistors of the complementary symmetry circuit are biased so that they conduct only when the signal voltage causes input current to flow in their base-emitter circuits. This action is diagrammed in Figure 13-4C. The positive “swing” of the applied signal voltage causes only the NPN transistor to conduct, while the negative “swing” of the applied signal voltage causes only the PNP transistor to conduct. With this connection, power efficiency is improved to better than 80%, significantly reducing the power dissipation required of the transistors.

Signal Levels and Distortion

The circuit of Figure 13-4B includes an input control potentiometer to set the level of the signal at the input to the multi-stage circuit. If the level is set too high, the signal at the collector of the transistor will try to “swing” beyond the limits of the power supply. The signal will not be able to exceed these limits, however, so the waveform will be clipped where it reaches V_{CC} and zero volts. Figure 13-5A shows the results of this effect. The resulting waveform distortion will produce unwanted harmonic and intermod frequencies. In practice, signal “swings” must be limited to about 80% of the power supply limits for minimal distortion. The percentage of distortion increases as the signals approach, then exceed the power supply limits.

Figures 13-5B and 13-5C show the effects of a different type of distortion that occurs with complementary-symmetry circuits. As you recall, the current flows in this circuit when the applied signal alternately causes the NPN and PNP transistors to conduct. The distortion occurs at the very low instantaneous voltages, when the waveform of the applied signal crosses zero as it changes phase from positive to negative, and vice-versa. At these points in the waveform, neither the NPN nor PNP transistors will conduct current, so there will be no output. This type of distortion is called “crossover distortion.”

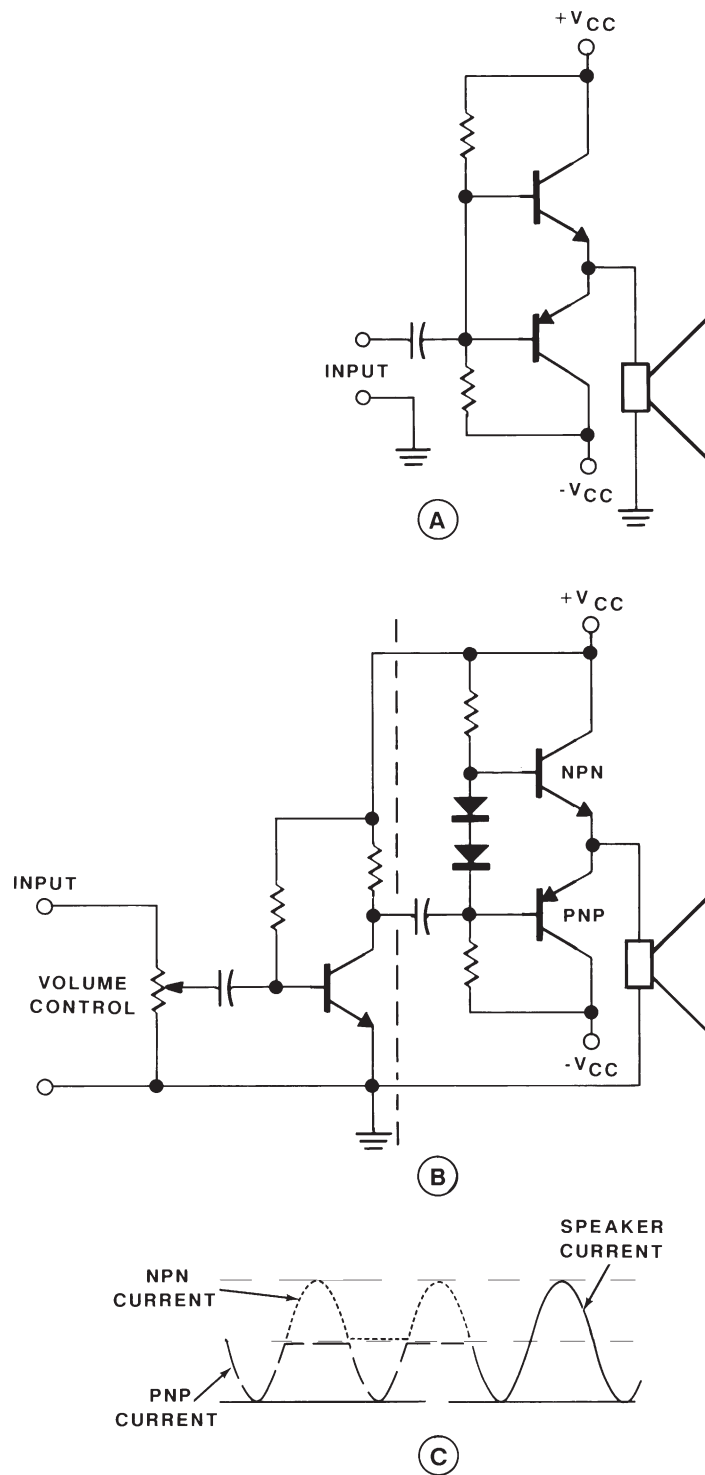
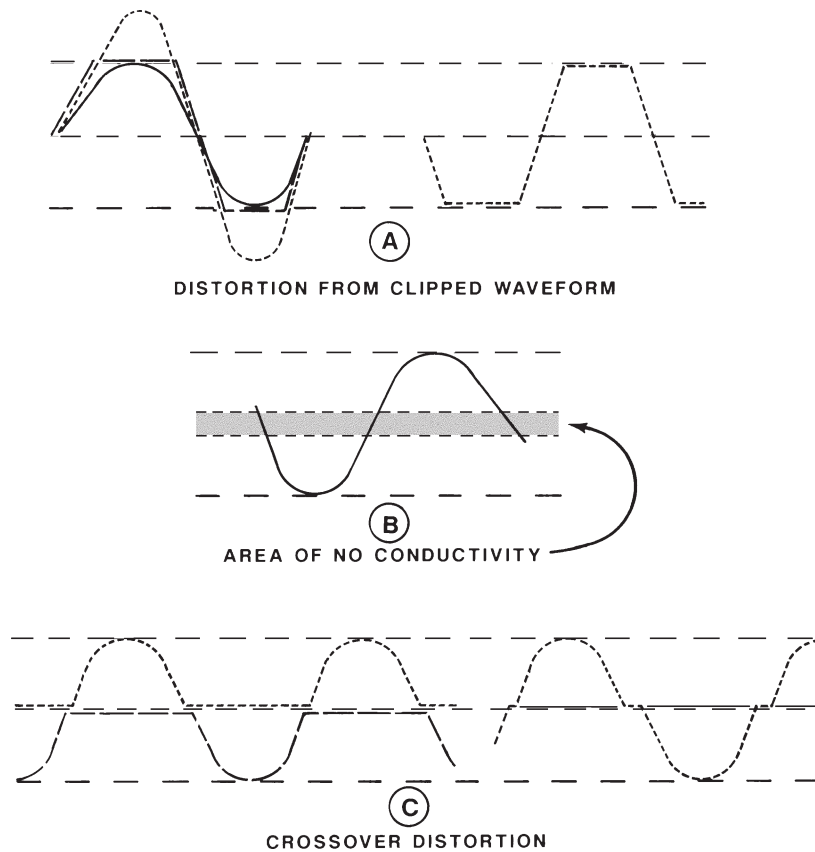


Figure 13-4
Complementary-symmetry power amplifier circuit.

**Figure 13-5**

Output waveform distortion.

The addition of the two diodes in the input section of the complementary-symmetry amplifier circuit of Figure 13-4B greatly reduce crossover distortion. The diodes provide a small DC voltage across the two input base terminals to keep both transistors conducting during the zero-crossing of the applied signal.

From your earlier studies of this course, you should recall that any waveform distortion produces harmonic and intermodulation frequencies, which are sometimes desirable. In audio amplifiers, however, such distortion is never desirable! The output waveforms of audio amplifiers must closely follow those of the input signals applied to them.

Amplifying Low-level Signals

It is very often necessary to amplify very low-level audio signals, such as those from low-sensitivity microphones. Amplifiers of low level signals must have high gain (amplification) factors to raise minute signals to levels large enough to drive loudspeakers. With such high gain factors, these amplifiers can also raise minute levels of noise, and even power-supply “hum”, to levels large enough to drive loudspeakers. Unless some circuit alternatives are applied, the results of high gain will include “noise” within the amplified signals.

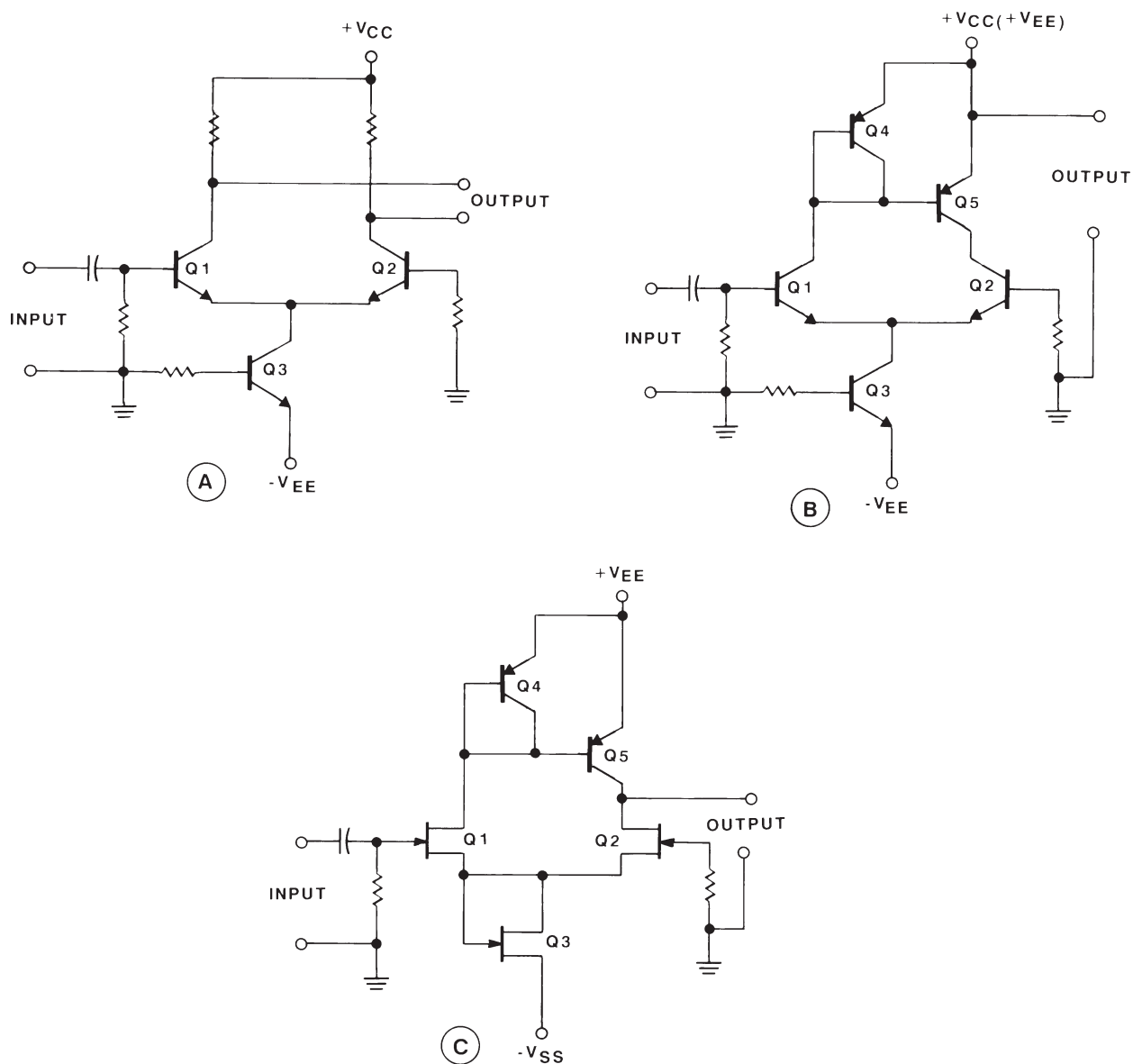
One very effective circuit technique that reduces the amplification of unwanted signals uses a three-transistor circuit arrangement as shown in Figure 13-6A. This circuit arrangement is called the “differential amplifier”. The circuit is balanced (like a balanced bridge circuit), so the output signal is obtained from between the collectors of transistors Q1 and Q2. The emitters of those transistors are connected to a common source of current through the collector of Q3.

When there is no input signal applied to this circuit, virtually all of the noise and hum appear equally in both transistors, Q1 and Q2. There is no difference in these signals at the collectors of Q1 and Q2, so there will be no signal at the output of the circuit between these terminals. All signals that are common to both transistors will not appear at the circuit’s output. This property of the differential amplifier circuit is called “common-mode rejection”.

With a signal applied to only one of the base inputs, such as the base of Q1, the transistors are no longer balanced to that signal. The amplified waveform of the input signal is a differential signal that will thus accurately appear across the collector outputs of Q1 and Q2.

Once the low-level signal is amplified by the differential amplifier, it is often connected to a circuit called a “current mirror”. This circuit, shown connected to the collectors of Q1 and Q2, in Figure 13-6B, is composed of two directly connected transistors, Q4 and Q5. This circuit arrangement excludes common mode signals, but amplifies differential signals. The circuit also provides a single-ended output signal line at the collector of Q5 that is referenced to ground.

Figure 13-6C shows a differential amplifier circuit that uses FETs instead of bipolar transistors. The input impedance of FETs is much higher than that of bipolars. This circuit is most appropriately used as an input amplifier for a low-level, high-impedance electret condenser microphone.

**Figure 13-6**

Transistor input amplifier circuit.

The Complete Amplifier

Figure 13-7 shows the entire multi-stage circuit of the audio amplifier which uses discrete bipolar transistors to amplify low-level signals from a microphone, and then raise signals to power levels required to drive a loudspeaker. This circuit is only representative of thousands of variations you may find in audio and telephone systems. However, it uses a variety of bipolar and field-effect transistors in circuit arrangements typical of those used in actual systems.

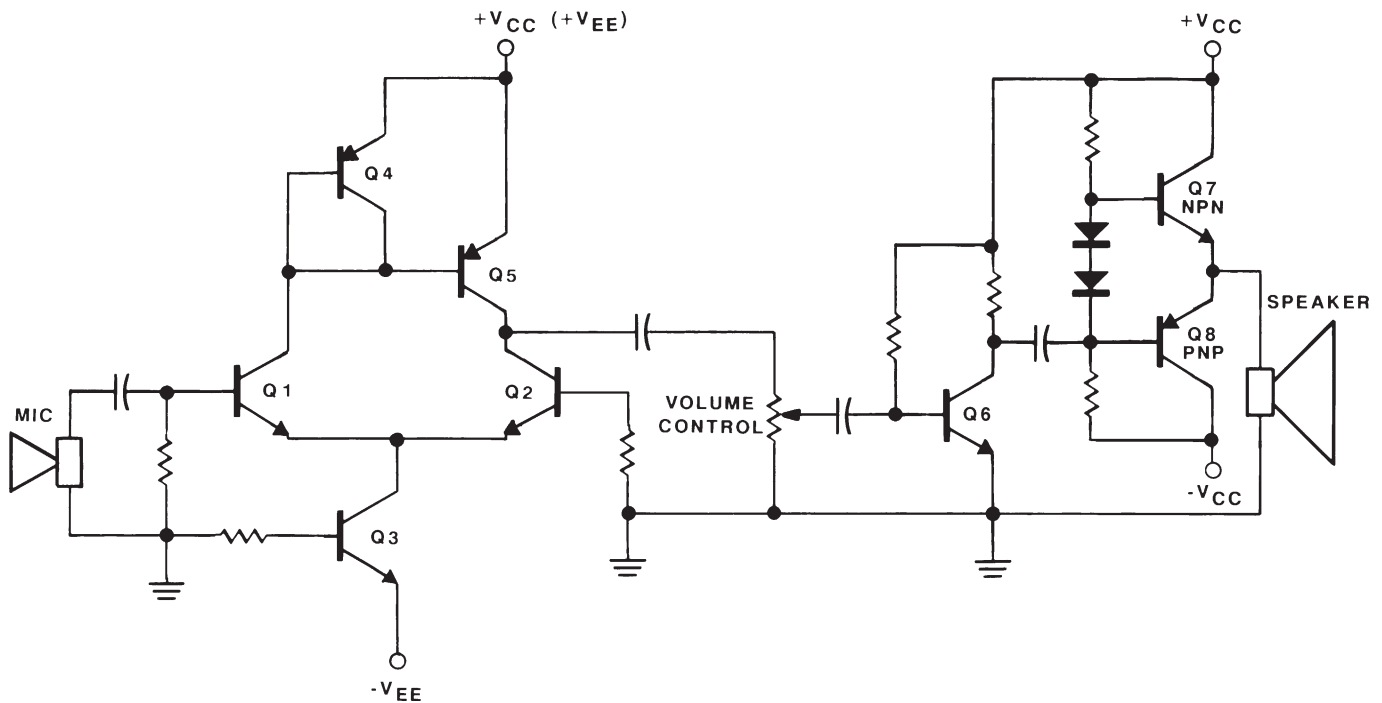


Figure 13-7
Complete audio amplifier.

The approximate gain of each of the circuit stages depends upon variables of the actual transistors selected, and the values of the resistors that surround the transistors. The precise calculation of these values is not the purpose of this course. However, we can generalize the gain of such circuits in order to obtain some perspective on their overall amplification function.

1. As we described earlier, common-collector circuits are used for the complementary-symmetry output stage of Figure 13-7 to drive the loudspeaker. The circuit of Q7 and Q8 has a voltage gain slightly less than one, and a current gain typically between 50 and 100. We'll call this voltage gain A_{v1} .
2. The common-emitter voltage amplifier, of Q6, drives the output circuit. Its voltage and current gain factors are typically a minimum of 30. The level (volume) control at the input to this stage can be set to assure a desired output signal level. We'll call this voltage gain A_{v2} .
3. The differential amplifier and current-mirror circuits of Q1 through Q5 have a nominal combined voltage and current gain typically a minimum of 60, about twice the gain of the single common-emitter amplifier circuit. We'll call this voltage gain A_{v3} .

We can now evaluate the entire amplifier, assuming a 12-volt DC power supply, and about 2.25 watts of signal output from the speaker, per our earlier calculations of power capacity. The overall voltage gain of the amplifier is the total of each of its stages: 0.98 for the output, 30 for the common-emitter amplifier, and 60 for the differential amplifier. The overall gain is:

$$\begin{aligned} A_{v\text{total}} &= A_{v1} \times A_{v2} \times A_{v3} \\ &= 0.98 \times 30 \times 60 = 1700 \end{aligned}$$

If we assume a 12-volt p-p output voltage “swing” across the 8-ohm speaker, we can determine the maximum voltage level required from the microphone at its loudest input level. This input signal level is determined as:

$$\begin{aligned} V_{\text{in}} &= V_{\text{out}} \div A_v \\ &= 12 \text{ Vp-p} \div (1700) = 7 \text{ mVp-p} \\ V_{\text{rms}} &= V_{\text{p-p}} \div 2.828 = 7 \div 2.828 \approx 2.5 \text{ mV} \end{aligned}$$

The maximum signal voltage from the microphone (in rms units, per “mic” specifications) is about 2.5 millivolts into an open circuit, or about 1.3 millivolts into a matched impedance. This signal from the microphone will yield the loudest sound level from the loudspeaker.

We can determine microphone requirements for this system, by referring back to Chapter 12. For this exercise, we’ll assume that the input “mic” is referenced to the louder level, 0 dB = 1 volt/10 microbar (84 dB SPL). We’ll also assume that the loudest sound applied to the microphone is 94 dB SPL, 10 dB above the “mic’s” sensitivity reference.

With a matched input impedance, a signal of 1.3 millivolts to the amplifier will drive the loudspeaker to full volume when the level control (P2) is at its highest setting. We will use this input signal voltage level to determine the minimum microphone sensitivity required for the audio system. The calculation is made by transposing the formula we used earlier in specifying microphone sensitivity:

$$\begin{aligned} (V_{\text{out}}) &= \log_{-1}(\text{dB}/20); \text{ therefore,} \\ \text{dB}_{\text{sens}} &= 20 \log (V_{\text{out}}) - (\text{dB}_{\text{max}} - \text{dB}_{\text{ref}}) \end{aligned}$$

The 1.3 millivolt signal is referenced for a dB_{max} of 94 dB SPL, while the sensitivity specification for the microphone is referenced for an 84 dB SPL level. The calculation is as follows:

$$\begin{aligned}\text{dB}_{\text{sens}} &= 20 \log(V_{\text{out}}) - (\text{dB}_{\text{max}} - \text{dB}_{\text{ref}}) \\ &= 20 \log(1.3 \times 10^{-3}) - (94 - 84) \approx -68 \text{ dB}\end{aligned}$$

A “mic” with a sensitivity as low as -68 dB can be used with this amplifier. Most microphones have higher sensitivity ratings, so their levels can be reduced with control potentiometers P1 and P2 for optimum performance with this amplifier.

Your study of the last few pages should help you understand the procedures used to determine amplifier requirements in audio and telecommunications systems. The amplifiers are used primarily to raise the level of a low input signal to a higher level required for communication. The process must usually consider the electrical factors of power, impedance, voltage, and gain.

INTEGRATED CIRCUIT (IC) AUDIO AMPLIFIERS

In the previous section of this course we reviewed the operation of amplifier circuits that used discrete (individual) transistors in combination to achieve specific circuit performance. We will now proceed to describe electronic devices that “integrate” these combination circuits within a single microminiature component. These devices are aptly called “integrated circuits”, and are often referred-to in many ways such as “ICs”, “IC devices”, “IC chips”, “microchips”, or “microcircuits”.

ICs are electronic devices that integrate a combination of discrete transistor circuits within a single “microchip” component.

Two general classifications of IC devices are available for audio amplification. These are operational amplifiers and audio power amplifiers. The operational amplifiers, often called “op-amps”, usually have differential inputs and lower power capacities, while the more specialized audio power amps often have single-ended inputs, with higher power capacities. The functions of the complete circuitry, as shown in Figure 13-7, are representative of a single IC device, with higher gain and performance, and with much lower cost, than the equivalent discrete circuit. So, while it is useful to understand electronic circuit functions by studying discrete circuits, modern electronic systems seldom use discrete devices in their circuitry.

Figure 13-8A shows the general schematic symbol for an op-amp, and Figures 13-8B and 13-8C show the entire simple circuitry required for using this IC as an audio amplifier. Figure 13-8D shows the typical internal circuitry of the op-amp as similar to the discrete circuitry of Figure 13-7.

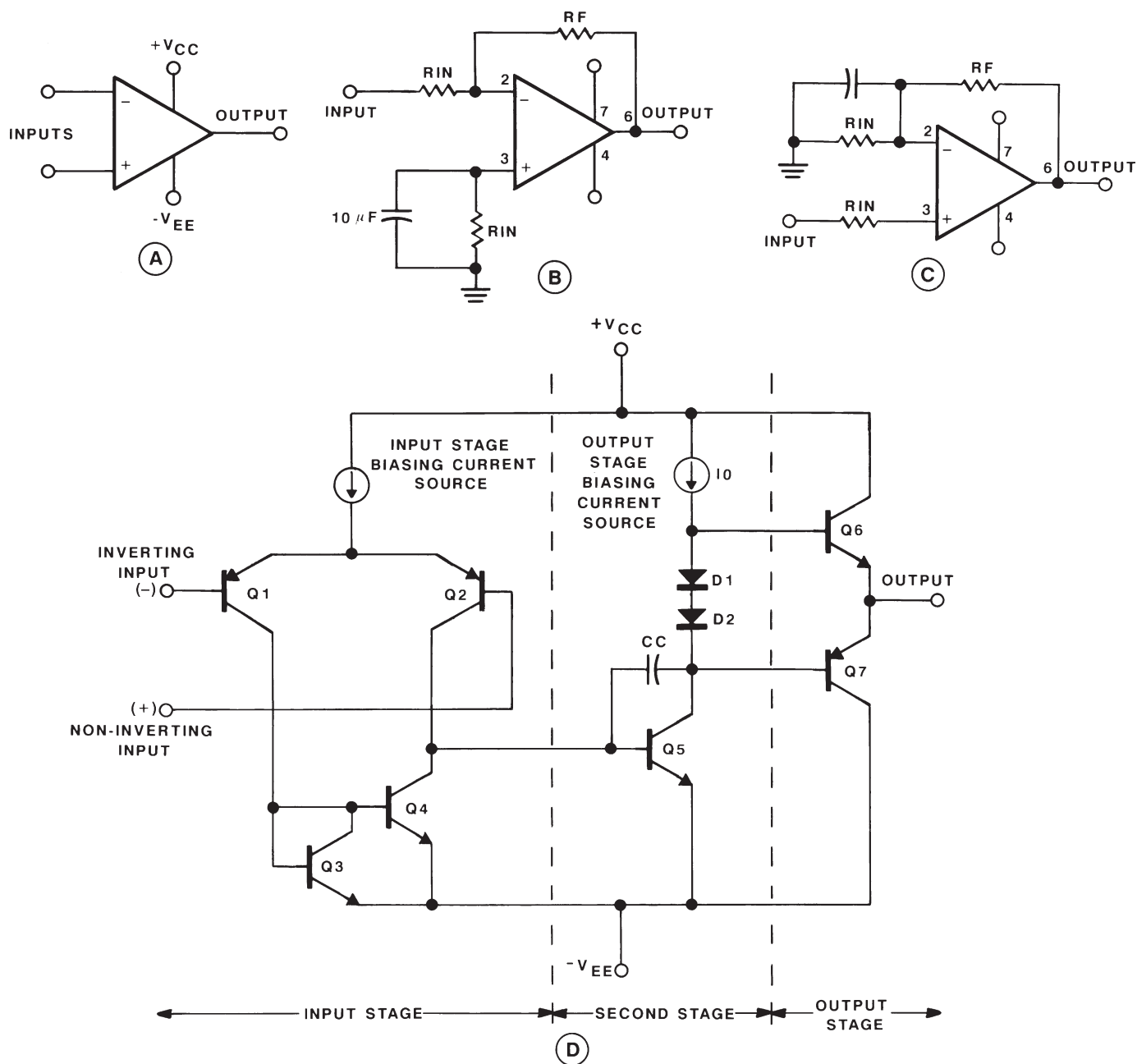


Figure 13-8
Integrated circuit operational amplifier.

The differential input terminals of an op-amp are labeled (–) and (+), to reference the phase of the output signal. The (+) input terminal is in-phase with the output signal, and it is called the “non-inverting input”. The (–) input terminal is 180° out-of-phase with the output signal, and it is called the “inverting input”. One of the features of the op-amp circuit is the availability of both inputs for a variety signal processing functions.

In operation, a portion of the output signal is “fed back” through R_F to the inverting input of the op-amp to control the gain of the circuit. The voltage gain of the inverting connection of the amplifier, per Figure 13-8B is simply the ratio of R_F to R_{IN} . With typical values of R_{IN} from 1 kilohm to 10 kilohms, voltage gains of any value, typically up to 104 at audio frequencies, can be obtained by selecting the correct value for R_F . The two forms of the formulas for calculating the gain of the inverting amplifier are:

$$A_V = \frac{R_F}{R_{IN}}; \text{ and } R_F = A_V R_{IN}$$

For a voltage gain of 5000, and an input resistance (R_{IN}) of 1 kilohm, such as obtained with the discrete circuit of Figure 13-7, we can calculate the value of R_F as follows:

$$R_F = A_V R_{IN} = 5000 \times 1 \text{ k}\Omega = 5 \times 10^6 \Omega = 5 \text{ M}\Omega$$

The voltage gain of the non-inverting connection of the amplifier, per Figure 13-8C is the ratio of R_F to R_{IN} plus One. As with the inverting amplifier, typical values of R_{IN} range from 1 kilohm to 10 kilohms, and voltage gains of any value to about 10^4 can be obtained by selecting the correct value for R_F . The two forms of the formulas for calculating the gain of the non-inverting amplifier are:

$$A_V = 1 + \frac{R_F}{R_{IN}}; \text{ and}$$

$$R_F = R_{IN}(A_V - 1)$$

For a voltage gain of 50 and an input resistance (R_{IN}) of 10 kilohms, we can calculate the value of R_F for the non-inverting amplifier of Figure 13-8C as follows:

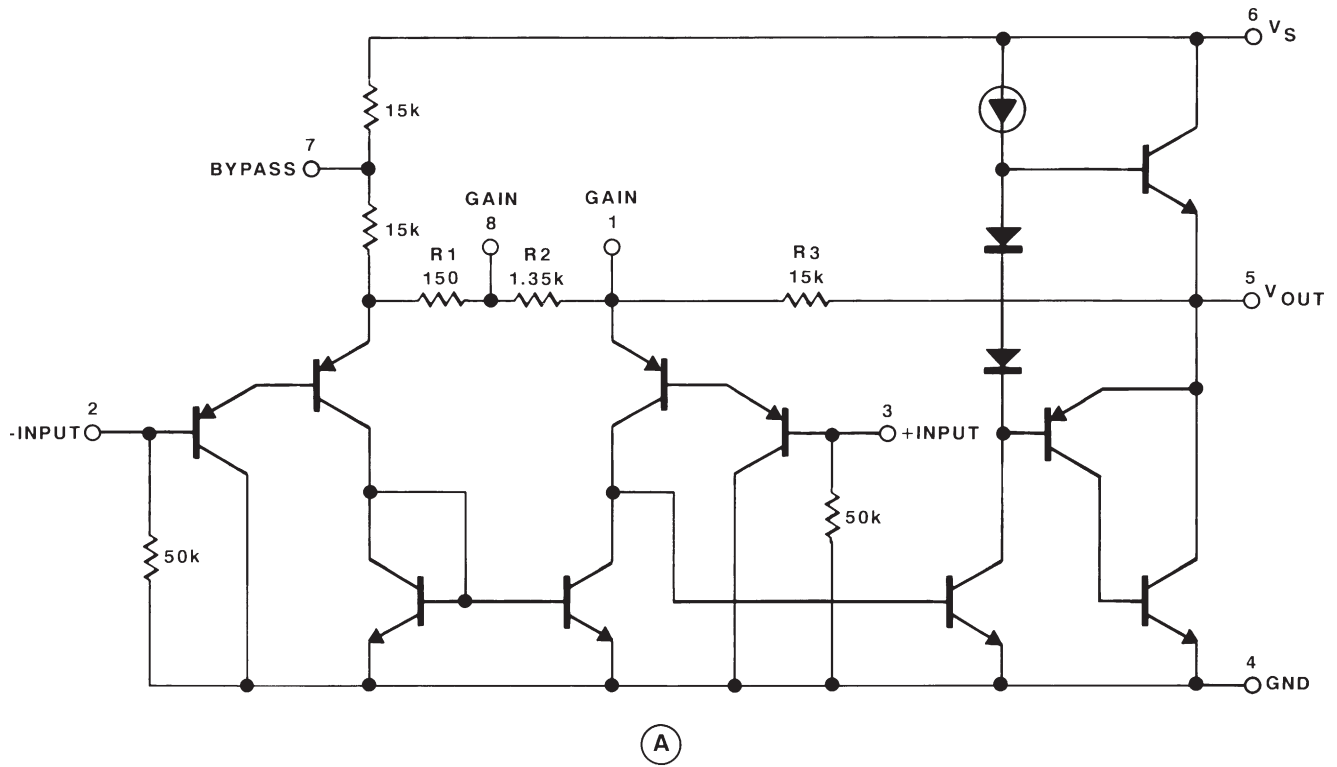
$$R_F = R_{IN}(A_V - 1) = 10 \text{ k}\Omega (50 - 1)$$

$$R_F = 4.9 \times 10^5 \Omega = 490 \text{ k}\Omega$$

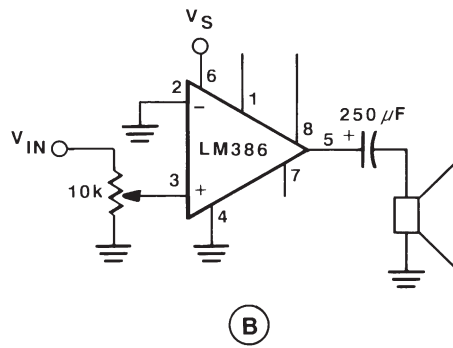
Figure 13-9A shows the schematic diagram of an IC audio amplifier circuit. This circuit is virtually identical to the op-amp, but it has lower gain limits and higher power capacity. The IC is intended to be used in a single-ended input circuit as shown in Figure 13-9B. It is simply a non-inverting op amp with internally-connected feedback resistance. Resistor R_3 forms the feedback circuit (R_F), with R_1 and R_2 as the input resistors (R_{IN}) for the gain circuit. Both resistors, R_1 and R_2 , are active in the circuit of Figure 13-9B, to provide a voltage gain of 20 for the circuit. Shorting the signal across R_2 with the 10 microfarad capacitor, as shown in Figure 13-9C, increases the voltage gain by a factor of 10, from 20 to 200.

PASSIVE COMPONENTS

Electronic circuits are comprised of active components (devices), such as transistors and ICs, and passive components, such as resistors, capacitors, and inductors. The resistors used in amplifier circuits generally function to set the correct “bias” operation of the active devices, and to set the gain of the circuits. The capacitors and inductors are frequency-selective components that generally function to set the frequency response of the circuits. For audio applications, the capacitors are smaller and less-costly than inductors, so capacitors are used more often than inductors for setting frequency response.



AMPLIFIER WITH GAIN=20



AMPLIFIER WITH GAIN=200

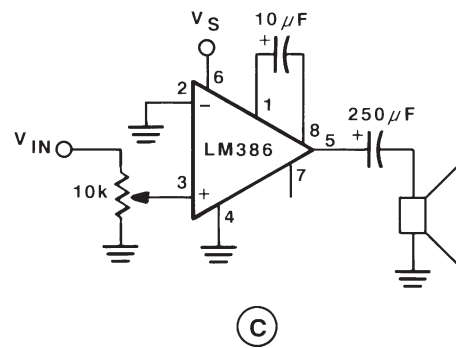
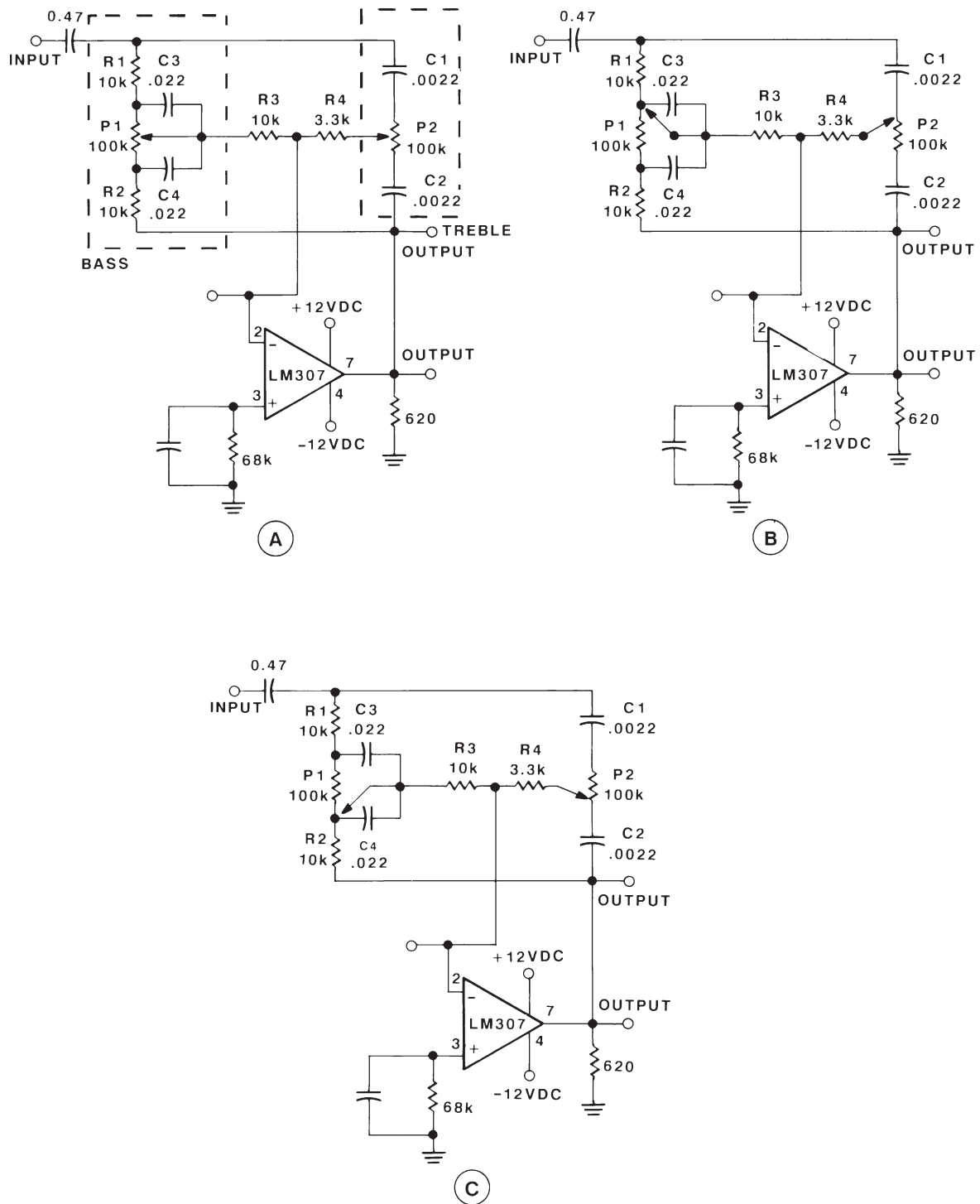


Figure 13-9

Capacitors are also used to remove signals from circuit paths by bypassing them to ground. They are used to block DC, but couple signals from one circuit path to another. In the circuit of Figure 13-8B, the 10 microfarad capacitor is used to bypass any signals present on pin 3 to ground, and assure that only DC voltage is present at that terminal. In the circuit of Figure 13-9C, the 250 microfarad capacitor is used to couple the output signal to the loudspeaker, but block the DC voltage on pin 5 from reaching the speaker.

Capacitors are used to provide more or less gain at different frequencies to set the frequency response of circuits. Figure 13-10A shows a typical bass boost/cut, treble boost/cut circuit used in the tone controls of audio amplifiers. The active device in this circuit is an op-amp. The input signal is applied to the non-inverting input of the op-amp, through a complicated RC network that is also a part of the feedback circuit. The circuit path through the 0.0022 microfarad capacitors (at the right side of the circuit drawing) provides for high-frequency (“treble”) boost and cut, while the circuit path through the 0.022 microfarad capacitors (at the left side of the circuit drawing) provides for low-frequency (“bass”) boost and cut.

Figure 13-10B shows the controls set for boost. The higher-frequency input signals couple through 0.0022 microfarad capacitor C1, and 3.3 kilohm resistor R4, to the amplifier’s inverting input. The feedback resistance, set in potentiometer P2, is 100 kilohms, so the gain of the amplifier is high for the higher frequencies.

**Figure 13-10**

Frequency compression circuits.

The lower-frequency input signals pass through R1 and R3, with 10 kilohms + 10 kilohms of input resistance to the op-amp's inverting input. The feedback resistance path is through R2 of 10 kilohms and potentiometer P1, set for 100 kilohms. The gain is high for the lower-frequency signals. The higher frequency signals in this signal path bypass P1, eliminating the higher gain it provides at the lower frequencies.

When the controls are set for bass and treble cut, as shown in Figure 13-10C, the ratios of input-to-feedback resistance and low and high frequencies are reduced. The bass circuit has an input resistance of 120 kilohms, with a feedback resistance of only 20 kilohms, while the treble circuit has an input resistance of 103.3 kilohms, with a feedback resistance of only 3.3 kilohms. The capacitors in the circuit couple and bypass resistance values at the low and high frequencies.

Replacing controls P1 and P2 with fixed resistors provides for frequency compensation of circuits, tailoring them to amplify some frequencies more than others. Such circuits are used extensively in telephone systems to limit their frequency response from 300 Hz to 3300 Hz. The circuits are also used to shape the frequency response of non-uniform components, such as microphones. In such applications, the circuits are said to provide an "equalization" function.

SUMMARY

Electronic amplifier circuits are used to raise the low signals levels from input transducers to the high power levels required to drive output transducers. These circuits function most efficiently when impedance factors are considered in their design. Low-impedance output amplifier circuits are required to drive loudspeakers to high power levels; and higher-impedance input and inter-stage circuits to greatly amplify the voltage levels of feeble signals.

Electronic amplifiers are comprised of active electronic devices, including transistors and integrated circuits. The variety of transistor types and circuit connections allows them to be used in both low- and high-impedance, and low- and high-gain circuits. The ICs combine these circuit connections within a single "chip" to provide for efficiencies in cost, size, and performance.

The IC devices are generally configured as operational amplifiers that have high input and low output impedances. They require simple combinations of resistor values to obtain needed voltage and power gain factors. IC devices, classified as audio amplifiers, have higher output power levels, and are often used to drive loudspeakers. They are fundamentally lower-gain versions of operational amplifiers, often with built-in resistor values to adjust for optimum operation.

Most electronic amplifiers use external capacitors to obtain specific frequency response requirements. Connecting these capacitors within the feedback loops of operational amplifiers can vary the exact degree of frequency vs. gain variation necessary for effective communication.

SELF-TEST REVIEW

1. List the ten common properties that define the way amplifiers process signals.

2. Which transistor circuit connection has the highest input impedance and lowest output impedance?
 - A. Common emitter.
 - B. Common collector.
 - C. Common source.
 - D. Common drain.
3. Which transistor type requires a positive voltage on two of its three terminals for operation?
 - A. NPN bipolar transistor.
 - B. PNP bipolar transistor.
 - C. N-channel FET.
 - D. P-channel FET.
4. In which transistor circuit connection is the output signal obtained from the emitter terminal?
 - A. Common emitter.
 - B. Common collector.
 - C. Common source.
 - D. Common drain.

5. Which power supply voltage and load impedance requires the highest power capacity of a transistor or IC device?
 - A. 20 volts, 12 ohms.
 - B. 18 volts, 10 ohms.
 - C. 16 volts, 8 ohms.
 - D. 12 volts, 4 ohms.

6. If the power supply voltage to a transistor is increased from 8 VDC to 12 VDC, by what factor will the transistor's power capacity increase?
 - A. 1.5
 - B. 2.0
 - C. 2.25
 - D. 3.0

7. The highest voltage gain and power capacity can be obtained by connecting which two stages together?
 - A. A common-emitter driving a common-emitter.
 - B. A common-collector driving a common-collector.
 - C. A common-emitter driving a common-collector.
 - D. A common-collector driving a common-emitter.

8. What is the major trade-off advantage of the two-transistor complementary-symmetry circuit over the single-transistor common-collector circuit?
 - A. Voltage gain.
 - B. Current gain.
 - C. Power capacity.
 - D. Power efficiency.

9. What is the name given to the waveform distortion that occurs when the signal swings beyond the limits of a device's power supply?
 - A. Crossover.
 - B. Clipping.
 - C. Harmonic.
 - D. Intermodulation.

10. What is the name given to the waveform distortion that occurs when the signal crosses zero volts driving a complementary symmetry output circuit?
 - A. Crossover.
 - B. Clipping.
 - C. Harmonic.
 - D. Intermodulation.
11. What is the major advantage of the differential amplifier in amplifying low-level signals?
 - A. Low distortion.
 - B. High gain.
 - C. Low noise.
 - D. High power capacity.
12. Common mode signals within the differential amplifier are:
 - A. rejected.
 - B. distorted.
 - C. amplified.
 - D. filtered.
13. The typical voltage gain of the differential amplifier circuit is about:
 - A. 5.
 - B. 50.
 - C. 500.
 - D. 5000.
14. The typical voltage gain of the complete amplifier circuit of Figure 4-16 is about:
 - A. 200.
 - B. 2000.
 - C. 20,000.
 - D. 200,000.
15. IC operational amplifiers have:
 - A. single-ended inputs and single-ended outputs.
 - B. single-ended inputs and differential outputs.
 - C. differential inputs and single-ended outputs.
 - D. differential inputs and differential outputs.

16. The (–) input to the op-amp indicates that:
 - A. it is out-of-phase with the output signal.
 - B. it is a differential input.
 - C. the input signal is subtracted from the output signal.
 - D. current flows only during the negative swing of the input signal.
17. The voltage gain of the operational amplifier depends upon the ratio of the feedback resistor to the:
 - A. output load resistor.
 - B. input source resistor.
 - C. internal source resistance of the IC.
 - D. internal load resistance of the IC.
18. Compared to op-amps IC audio amplifiers generally have:
 - A. higher gain and higher power capacity.
 - B. higher gain and lower power capacity.
 - C. lower gain and higher power capacity.
 - D. lower gain and lower power capacity.
19. Placing a capacitor across the feedback resistor in the op-amp circuit of Figure 13-8B will:
 - A. increase the circuit gain at low frequencies.
 - B. increase the circuit gain at high frequencies.
 - C. decrease the circuit gain at low frequencies.
 - D. decrease the circuit gain at high frequencies.
20. With a 8 VDC power supply, and a collector impedance of 400 ohms, what is the power capacity of a transistor audio amplifier stage?
_____ mW
21. With a 12-VDC power supply, what is the maximum power available to a 4-ohm loudspeaker, driven by a matched impedance common-collector transistor audio amplifier stage? _____ watts
22. What is the DC power required from the 12-VDC supply of Problem 21? _____ W
23. A microphone has a maximum rms output voltage of 1 millivolt. It is desired to raise that signal to drive a 4-ohm loudspeaker to maximum power with a 12 VDC power supply. What is the total voltage gain required of the amplifier circuit? _____

24. The microphone in Question 23 is driven with a sound power level of 90 dB SPL. It is referenced to 74 dB SPL. What is the minimum sensitivity specification required of the microphone? _____ dB
25. An inverting-input op-amp circuit has a voltage gain of 800. Its input resistor is 1 kilohm. What is the value of its feedback resistor? _____ ohms
26. A non-inverting op-amp circuit has an input resistor of 1 kilohm and a feedback resistor of 15 kilohms. What is the circuit's voltage gain?

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